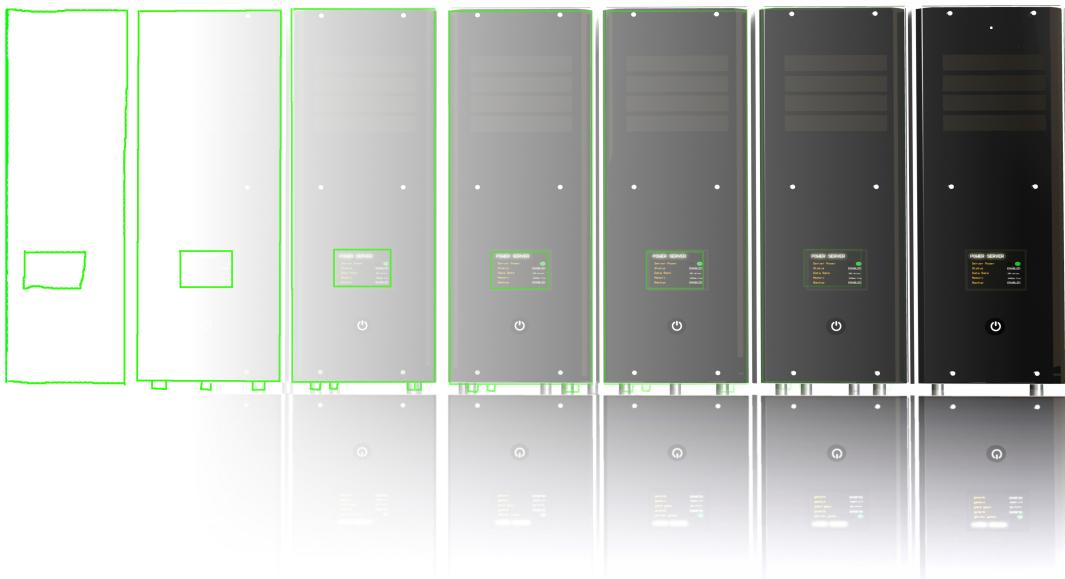


Operating Systems

Principles and Practice



Anderson and Dahlin
v. 0.22

Operating Systems: Principles and Practice

Version 0.22

Base revision e8814fe, Fri Jan 13 14:51:02 2012 -0600.

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Preface

Why We're Writing This Book

There has been a huge amount of innovation in both the principles and practice of operating systems over the past two decades. The pace of innovation in operating systems has, if anything, increased over the past few years, with the introduction of the iOS and Android operating systems for smartphones, the shift to multicore computers, and the advent of cloud computing.

Yet many operating systems textbooks treat the field as if it is static — that almost everything we need to cover in our classes was invented in the 60's and 70's. No! We strongly believe that students both need to, and can, understand modern operating systems concepts and modern implementation techniques. At Texas and Washington, we have been teaching the topics covered in this textbook for years, winning awards for our teaching. The approach in this book is the same one we use in organizing our own courses: that it is essential for students to learn both *principles* and *practice*, that is, both concepts and implementation, rather than either alone.

Although this book focuses on operating systems, we believe the concepts and principles are important for anyone getting a degree in computer science or computer engineering. The core ideas in operating systems — protection, concurrency, virtualization, resource allocation, and reliable storage — are widely used throughout computer science. Anyone trying to build resilient, secure, flexible computer systems needs to have a deep grounding in these topics and to be able to apply these concepts in a variety of settings. This is especially true in a modern world where nearly everything a user does is distributed, and nearly every computer is multi-core. Operating systems concepts are popping up in many different areas; even web browsers and cloud computing platforms have become mini-operating systems in their own right.

Precisely because operating systems concepts are among the most difficult in all of computer science, it is also important to ground students in how these ideas are applied in practice in real operating systems of today. In this book, we give students both concepts and working code. We have designed the book to support and be complemented with a rigorous operating systems course project,

such as Nachos, Pintos, JOS, or Linux. Our treatment, however, is general — it is not our intent to completely explain any particular operating system or course project.

Because the concepts in this textbook are so fundamental to much of the practice of modern computer science, we believe a rigorous operating systems course should be taken early in an undergraduate's course of study. For many students, an operating systems class is the ticket to an internship and eventually to a full-time position. We have designed this textbook assuming only that students have taken a class on data structures and one on basic machine structures. In particular, we have designed our book to interface well if students have used the Bryant and O'Halloran textbook on machine structures. Since some schools only get through the first half of Bryant and O'Halloran in their machine structures course, our textbook reviews and covers in much more depth the material from the second half of that book.

An Overview of the Content

The textbook is organized to allow each instructor to choose an appropriate level of depth for each topic. Each chapter begins at a conceptual level, with implementation details and the more advanced material towards the end. A more conceptual course will skip the back parts of several of the chapters; a more advanced or more implementation-oriented course will need to go into chapters in more depth. No single semester course is likely to be able to cover every topic we have included, but we think it is a good thing for students to come away from an operating systems course with an appreciation that there is still a lot for them to learn.

Because students learn more by needing to solve problems, we have integrated some homework questions into the body of each chapter, to provide students a way of judging whether they understood the material covered to that point. A more complete set of sample assignments is given at the end of each chapter.

The book is divided into five parts: an introduction (Chapter 1), kernels and processes (Chapters 2-3), concurrency, synchronization and scheduling (Chapters 4-7), memory management (Chapters 8-10), and persistent storage (Chapters 11-13).

The goal of chapter 1 is to introduce the recurring themes found in the later chapters. We define some common terms, and we provide a bit of the history of the development of operating systems.

Chapter 2 covers kernel-based process protection — the concept and implementation of executing a user program with restricted privileges. The concept of protected execution and safe transfer across privilege levels is a key concept to most modern computer systems, given the increasing salience of computer security issues. For a quick introduction to the concepts, students need only

read through 2.3.2; the chapter then dives into the mechanics of system calls, exceptions and interrupts in some detail. Some instructors launch directly into concurrency, and cover kernels and kernel protection afterwards, as a lead-in to address spaces and virtual memory. While our textbook can be used that way, we have found that students benefit from a basic understanding of the role of operating systems in executing user programs, before introducing concurrency.

Chapter 3 is intended as an impedance match for students of differing backgrounds. Depending on student background, it can be skipped or covered in depth. The chapter covers the operating system from a programmer’s perspective: process creation and management, device-independent input/output, interprocess communication, and network sockets. Our goal is that students be able to understand at a detailed level what happens between a user clicking on a link in a web browser, and that request being transferred through the operating system kernel on each machine to the web server running at user-level, and back again. The second half of Chapter 3 dives into the organization of the operating system itself — how device drivers and the hardware abstraction layer work in a modern operating system; the difference between a monolithic and a microkernel operating system; and how policy and mechanism can be separated in modern operating systems.

Chapter 4 motivates and explains the concept of threads. Because of the increasing importance of concurrent programming, and its integration with Java, many students will have been introduced to multi-threaded programming in an earlier class. This is a bit dangerous, as testing will not expose students to the errors they are making in concurrent programming. Thus, the goal of this chapter is to provide a solid conceptual framework for understanding the semantics of concurrency, as well as how concurrent threads are implemented in both the operating system kernel and in user-level libraries. Instructors needing to go more quickly can omit Section 3.4 and 3.5.

Chapter 5 discusses the synchronization of multi-threaded programs, a central part of all operating systems and increasingly important in many other contexts. Our approach is to describe one effective method for structuring concurrent programs (monitors), rather than to cover in depth every proposed mechanism. In our view, it is important for students to master one methodology, and monitors are a particularly robust and simple one, capable of implementing most concurrent programs efficiently. Implementation of synchronization primitives are covered in Section 5.5; this can be skipped without compromising student understanding.

Chapter 6 discusses advanced topics in concurrency, including deadlock, synchronization across multiple objects, and advanced synchronization techniques like read-copy-update (RCU). This is material is important for students to know, but most semester-long operating systems courses will only be able to briefly touch upon these issues.

Chapter 7 covers the concepts of resource allocation in the specific context of

processor scheduling. After a quick tour through the tradeoffs between response time and throughput for uniprocessor scheduling, the chapter covers a set of more advanced topics in affinity and gang scheduling, power-aware and deadline scheduling, as well as server scheduling, basic queueing theory and overload management.

Chapter 8 explains hardware and software address translation mechanisms. The first part of the chapter covers how to provide flexible memory management through multilevel segmentation and paging. Section 8.3 then considers how hardware makes flexible memory management efficient through translation lookaside buffers and virtually addressed caches, and how these are kept consistent as the operating system changes the addresses assigned to each process. We conclude with a discussion of modern software-based protection mechanisms such as those found in Android.

Chapter 9 covers caching and virtual memory. Caches are of course central to many different types of computer systems. Most students will have seen the concept of a cache in an earlier class machine structures, so our goal here is to cover the theory and implementation of caches: when they work and when they don't, and how they are implemented in hardware and software. While it might seem that we could skip virtual memory, many systems today provide programmers the abstraction of memory-mapped files, and these rely on the same mechanisms as in traditional virtual memory.

Chapter 10 discusses advanced topics in memory management. Address translation hardware and software can be used for a number of different features in modern operating systems, such as zero copy I/O, copy on write, process checkpointing, and recoverable virtual memory. As this is more advanced material, it can be skipped for time.

Chapter 11 sketches the characteristics of storage hardware, specifically block storage devices such as magnetic disks and flash memory. The last two decades have seen rapid change in storage technology affecting both application programmers and operating systems designers; this chapter provides a snapshot for students, as a building block for the next two chapters. Classes in which students have taken a computer architecture course that covers these topics may choose to skip this chapter.

Chapter 12 uses file systems as a case study of how complex data structures can be organized on block storage devices to achieve flexibility and performance.

Chapter 13 explains the concept and implementation of reliable storage, using file systems as a concrete example. Starting with the ad hoc techniques in UNIX fsck for implementing a reliable file system, the chapter explains checkpointing and write ahead logging as alternate implementation strategies for building reliable storage, and it discusses how redundancy such as checksums and replication are used to improve reliability and availability.

We are contemplating adding several chapters on networking and distributed

operating systems topics, but we are still considering what topics we can reasonably cover. We will be developing this material over the coming months.

Chapter 1

Introduction

“Everything I need to know I learned in kindergarten.” – Robert Fulgham

How do we construct reliable, portable, efficient and secure computer systems? An essential component is the computer’s operating system — the software that manages a computer’s resources.

First, the bad news: operating systems concepts are among the most complex topics in computer science. A modern general-purpose operating system can run to over 50 million lines of code, or in other words, more than a thousand times as long as this textbook. New operating systems are being written all the time. If you are reading this textbook on an e-book reader, tablet, or smartphone, there is an operating system managing the device. Since we will not be able to cover everything, our focus will be on the essential concepts for building computer systems, ones that every computer scientist should know.

Now the good news: operating systems concepts are also among the most accessible topics in computer science. Most of the topics in this book will seem familiar to you — if you have ever tried to do two things at once, or picked the wrong line at a grocery store, or tried to keep a roommate or sibling from messing with your things, or succeeded at pulling off an April Fool’s joke. Each of these has an analogue in operating systems, and it is this familiarity that gives us hope that we can explain how operating systems do their work in a single textbook. All we will assume of the reader is a basic understanding of the operation of a computer and the ability to read pseudo-code.

We believe that understanding how operating systems work is essential for any student interested in building modern computer systems. Of course, everyone who uses a computer or a smartphone or even a modern toaster uses an

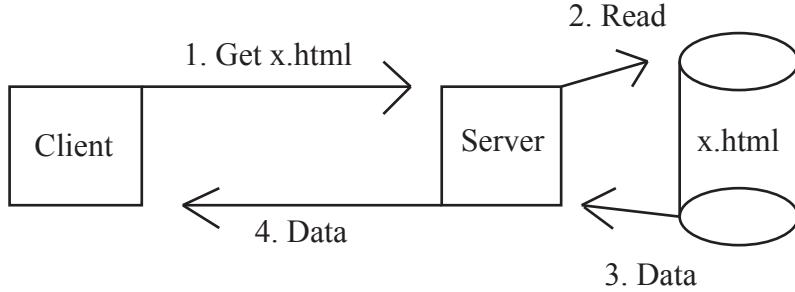


Figure 1.1: The operation of a web server.

operating system, so understanding the function of an operating system is useful to most computer scientists. Our goal in this book is to go much deeper than that, to explain the technologies used inside operating systems, technologies many of us rely on every day without realizing it.

Software engineers often encounter challenges similar to those faced by operating systems when building other complex systems, and they use many of the same technologies and design patterns. Whether your goal is to work on the internals of an operating system kernel, or to build the next generation of software for cloud computing, secure web browsers, game consoles, graphical user interfaces, media players, databases, or multicore software, the concepts and abstractions needed for reliable, portable, efficient and secure software are much the same. In our experience, the best way to learn these concepts is to study how they are used in operating systems, but we hope you will apply these concepts to a much broader range of computer systems.

To get started, consider the web server in Figure 1.1. Its behavior is amazingly simple: it receives a packet containing the name of the web page from the network. The web server decodes the packet, fetches the file from disk, and sends the contents back over the network to the user.

Part of an operating system's job is to make it easy to write applications like web servers. But if we dig a bit deeper, this simple story quickly raises as many questions as it answers:

- Many web requests involve both data and computation. For example, the Google home page presents a simple text box, but each search query entered in that box consults databases spread over literally thousands of machines. To keep their software manageable, web servers often invoke helper applications, e.g., to manage the actual search function. These helper applications need to communicate with the main web server for

this to work. How does the operating system enable multiple applications to communicate with each other?

- What if two users (or a million) try to request a web page from the server at the same time? A simple approach might be to handle each request in turn. If any individual request takes a long time, however, this approach would mean that everyone else would need to wait for it to complete. A faster, but more complex, solution is to *multitask*: to juggle the handling of multiple requests at once. Multitasking is especially important on modern multicore computers, as it provides a way to keep many processors busy. How does the operating system enable applications to do multiple things at once?
- For better performance, the web server might want to keep a copy, sometimes called a *cache*, of recently requested pages, so that the next user to request the same page can be returned the results from the cache, rather than starting the request from scratch. This requires the application to synchronize access to the cache's data structures by the thousands of web requests being handled at the same time. How does the operating system support application synchronization to shared data?
- To customize and animate the user experience, it is common for web servers to send clients scripting code, along with the contents of the web page. But this means that clicking on a link can cause someone else's code to run on your computer. How does the client operating system protect itself from being compromised by a computer virus surreptitiously embedded into the scripting code?
- Suppose the web site administrator uses an editor to update the web page. The web server needs to be able to read the file that the editor wrote; how does the operating system store the bytes on disk so that later on the web server can find and read them?
- Taking this a step further, the administrator probably wants to be able to make a consistent set of changes to the web site, so that embedded links are not left dangling, even temporarily. How can the operating system enable users to make a set of changes to a web site, so that requests either see the old pages or the new pages, but not a mishmash of the two?
- What happens when the client browser and the web server run at different speeds? If the server tries to send the web page to the client faster than the client can draw the page, where are the contents of the file stored in the meantime? Can the operating system decouple the client and server so that each can run at its own speed, without slowing the other down?
- As demand on the web server grows, the administrator is likely to want to move to more powerful hardware, with more memory, more processors, faster network devices, and faster disks. To take advantage of this new

hardware, does the web server need to be re-written from scratch, or can it be written in a hardware-independent fashion? What about the operating system — does it need to be re-written for every new piece of hardware?

We could go on, but you get the idea. This book will help you understand the answers to these questions, and more.

Goals of this chapter

The rest of this chapter discusses three topics in detail:

- **OS Definition.** What is an operating system and what does it do?
- **OS Challenges.** How should we evaluate operating systems, and what are some of the tradeoffs their designers face?
- **OS Past, Present and Future.** What is the history of operating systems, and what new functionality are we likely to see in future operating systems?

1.1 What is an operating system?

Definition: **operating system** An *operating system* is the layer of software that manages a computer's resources for its users and their applications. Operating systems run in a wide range of computer systems. Sometimes they are invisible to the end user, controlling embedded devices such as toasters, gaming systems, and the many computers inside modern automobiles and airplanes. Operating systems are also an essential component of more general-purpose systems such as smartphones, desktop computers, and servers.

Our discussion will focus on general-purpose operating systems, because the technologies they need are a superset of the technologies needed for embedded systems. Increasingly though, technologies developed for general-purpose computing are migrating into the embedded sphere. For example, early mobile phones had simple operating systems to manage the hardware and to run a handful of primitive applications. Today, smartphones — phones capable of running independent third party applications — are the fastest growing part of the mobile phone business. These new devices require much more complete operating systems, with sophisticated resource management, multi-tasking, security and failure isolation.

Likewise, automobiles are increasingly software controlled, raising a host of operating system issues. Can anyone write software for your car? What if the software fails while you are driving down the highway? How might the operating system of your car be designed to prevent a computer virus from

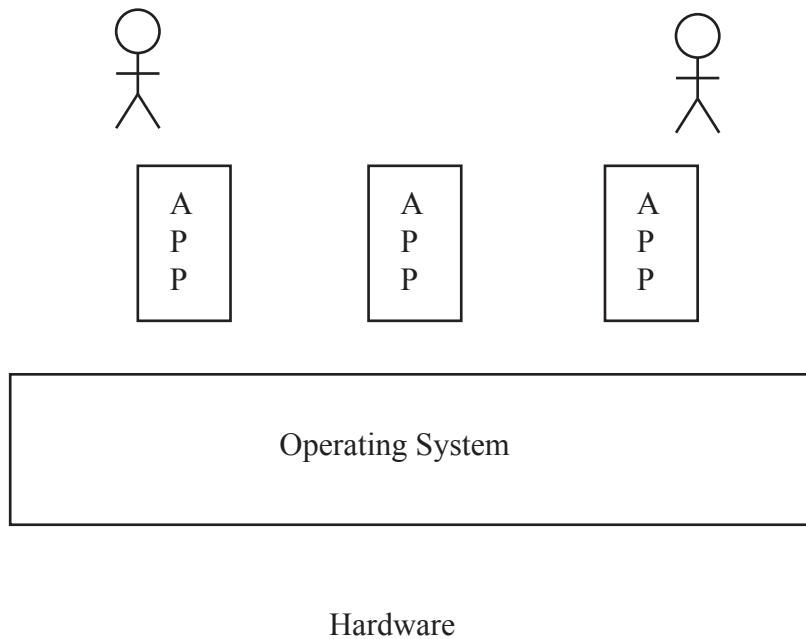


Figure 1.2: A general-purpose operating system

hijacking control of your car’s computers? Although this might seem far fetched, researchers recently demonstrated that they could remotely turn off a car’s braking system through a computer virus introduced into the car’s computers through a hacked car radio. A goal of this book is to explain how to build more reliable and secure computer systems in a variety of contexts.

For general-purpose systems, users interact with applications, applications execute in an environment provided by the operating system, and the operating system mediates access to the underlying hardware (Figure 1.2, and expanded in Figure 1.3). What do we need from an operating system to be able to run a group of programs? Operating systems have three roles:

- Operating systems play *referee* — they manage shared resources between different applications running on the same physical machine. For example, an operating system can stop one program and start another. Operating systems isolate different applications from each other, so that if there is a bug in one application, it does not corrupt other applications running on the same machine. The operating system must protect itself and other applications from malicious computer viruses. And since the applications

An Expanded View of an Operating System

Figure 1.3 shows the structure of a general-purpose operating system, as an expansion on the simple view presented in Figure 1.2. At the lowest level, the hardware provides processor, memory, and a set of devices for providing the user interface, storing data and communicating with the outside world. The hardware also provides primitives that the operating system can use to provide fault isolation and synchronization. The operating system runs as the lowest layer of software on the computer, with a device-specific layer interfaces to the myriad hardware devices, and a set of device-independent services provided to applications. Since the operating system needs to be able to isolate malicious and buggy applications from affecting other applications or the operating system itself, much of the operating system runs in a separate execution environment protected from application code. A portion of the operating system can also run as a library linked into each application. In turn, applications run in an execution context provided by the operating system. The application context is much more than a simple abstraction on top of hardware devices: applications execute in a virtual environment that is both more constrained (to prevent harm), more powerful (to mask hardware limitations), and more useful (via common services), than the underlying hardware.

are sharing physical resources, the operating system needs to decide which applications get which resources.

- Operating systems play *illusionist* — they provide an abstraction physical hardware to simplify application design. To write a “hello world” program, you do not need (or want!) to think about how much physical memory the system has, or how many other programs might be sharing the computer’s resources. Instead, operating systems provide the illusion of a nearly infinite memory, as an abstraction on top of a limited amount of physical memory. Likewise, operating systems provide the illusion that each program has the computer’s processors entirely to itself. Obviously, the reality is quite different! These illusions enable applications to be written independently of the amount of physical memory on the system or the physical number of processors. Because applications are written to a higher level of abstraction, the operating system is free to change the amount of resources assigned to each application as applications start and stop.
- Operating systems provide *glue* — a set of common services between applications. An important benefit of common services is to facilitate sharing between applications, so that, for example, cut and paste works uniformly across the system and a file written by one application can be read by another. Many operating systems provide a common set of user interface routines to help applications provide a common “look and feel.” Perhaps most importantly, operating systems provide a layer separating applica-

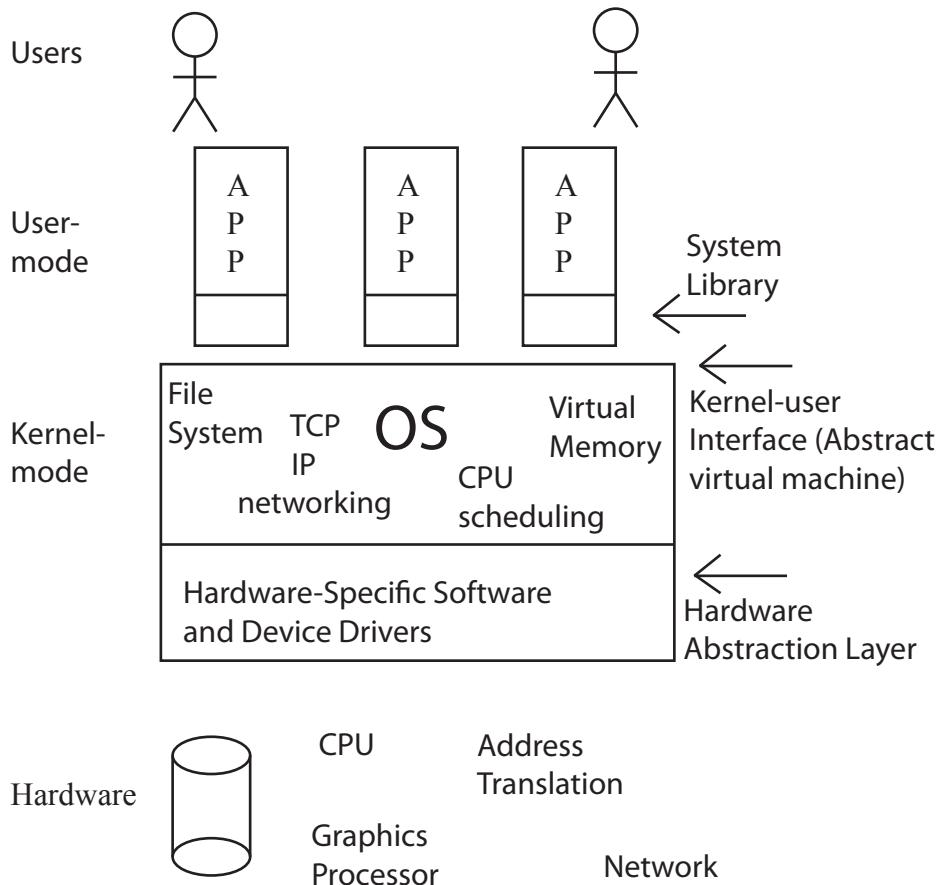


Figure 1.3: A general-purpose operating system: expanded view

tions from hardware input and output devices, so that applications can be written independently of which specific keyboard, mouse or disk drive is being used on a particular computer.

We next discuss these three roles in a bit more detail.

1.1.1 Resource sharing: Operating system as referee

Sharing is central to most uses of computers. Right now, my laptop is running a browser, podcast library, text editor, email program, document viewer, and newspaper. The operating system must somehow keep all of these activities separate, yet allow each the full capacity of the machine if the others aren't running. At a minimum, when one program stops running, the operating system should

let me run another. Better, the operating system should allow multiple applications to run at the same time, as when I read email while I am downloading a security patch to the system software.

Even individual applications can be designed to do multiple things at once. For instance, a web server will be more responsive to its users if it can handle multiple requests at the same time rather than waiting for each to complete before the next one starts running. The same holds for the browser — it is more responsive if it can start drawing a page while the rest of the page is still being transferred. On multiprocessors, the computation inside a parallel application can be split into separate units that can be run independently for faster execution. The operating system itself is an example of software written to be able to do multiple things at once. As we will describe later, the operating system is a customer of its own abstractions.

Sharing raises several challenges for an operating system:

- **Resource Allocation.** The operating system must keep all of the simultaneous activities separate, allocating resources to each as appropriate. A computer usually has only a few processors and a finite amount of memory, network bandwidth, and disk space. When there are multiple tasks to do at the same time, how should the operating system choose how many resources to give to each? Seemingly trivial differences in how resources are allocated can have a large impact on user-perceived performance. As we will see later, if the operating system gives too little memory to a program, it will not only slow down that particular program, it can dramatically hurt the performance of the entire machine.

As another example, what should happen if an application executes an infinite loop:

```
while(true){  
    ;  
}
```

If programs ran directly on the raw hardware, this code fragment would lock up the computer, making it completely non-responsive to user input. With resource multiplexing provided by the operating system, the specific application might lock up, but other programs can proceed unimpeded. Additionally, the user can ask the operating system to force the looping program to exit.

- **Isolation.** An error in one application should not disrupt other applications, or even the operating system itself. This is called *fault isolation*. Anyone who has taken an introductory computer science class knows the value of an operating system that can protect itself and other applications from programmer bugs. Debugging would be vastly harder if an error

Definition: **fault isolation**

in one program could corrupt data structures in other applications. Likewise, downloading and installing a screen saver or other application should not crash other unrelated programs, nor should it be a way for a malicious attacker to surreptitiously install a computer virus on the system. Nor should one user be able to access or change another's data without permission.

Fault isolation requires restricting the behavior of applications to less than the full power of the underlying hardware. Given access to the full capability of the hardware, any application downloaded off the web, or any script embedded in a web page, would have complete control of the machine. Thus, it would be able to install spyware into the operating system to log every keystroke you type, or record the password to every website you visit. Without fault isolation provided by the operating system, any bug in any program might cause the disk to become irretrievably corrupted. Erroneous or malignant applications would cause all sorts of havoc.

- **Communication.** The flip side of isolation is the need for communication between different applications and between different users. For example, a web site may be implemented by a cooperating set of applications: one to select advertisements, another to cache recent results, yet another to fetch and merge data from disk, and several more to cooperatively scan the web for new content to index. For this to work, the various programs need to be able to communicate with one another. If the operating systems is designed to prevent bugs and malicious users and applications from affecting other users and their applications, how does the operating system support communication to share results? In setting up boundaries, an operating system must also allow for those boundaries to be crossed in carefully controlled ways as the need arises.

In its role as a referee, an operating system is somewhat akin to that of a government, or perhaps a particularly patient kindergarten teacher, balancing needs, separating conflicts, and facilitating sharing. One user should not be able to hog all of the system's resources or to access or corrupt another user's files without permission; a buggy application should not be able to crash the operating system or other unrelated applications; and yet applications also need to be able to work together. Enforcing and balancing these concerns is the role of the operating system.

Exercises

Take a moment to speculate. We will provide answers to these questions throughout the rest of the book, but given what you know now, how would you answer them? Before there were operating systems, someone needed to develop solutions, without being able to look them up! How would you have designed the first operating system?

1. Suppose a computer system and all of its applications are completely bug free. Suppose further that everyone in the world is completely honest and trustworthy. In other words, we do not need to consider fault isolation.
 - a. How should the operating system allocate time on the processor? Should it give all of the processor to each application until it no longer needs it? If there are multiple tasks ready to go at the same time, should it schedule the task with the least amount of work to do or the one with the most? Justify your answer.
 - b. How should the operating system allocate physical memory between applications? What should happen if the set of applications do not all fit in memory at the same time?
 - c. How should the operating system allocate its disk space? Should the first user to ask be able to grab all of the free space? What would the likely outcome be for that policy?
2. Now suppose the computer system needs to support fault isolation. What hardware and/or operating support do you think would be needed to accomplish this goal?
 - a. For protecting an application's data structures in memory from being corrupted by other applications?
 - b. For protecting one user's disk files from being accessed or corrupted by another user?
 - c. For protecting the network from a virus trying to use your computer to send spam?
3. How should an operating system support communication between applications?
 - a. Through the file system?
 - b. Through messages passed between applications?
 - c. Through regions of memory shared between the applications?
 - d. All of the above? None of the above?

1.1.2 Mask hardware limitations: Operating system as illusionist

A second important role of operating systems is to mask the restrictions inherent in computer hardware. Hardware is necessarily limited by physical constraints — a computer has only a limited number of processors and a limited amount

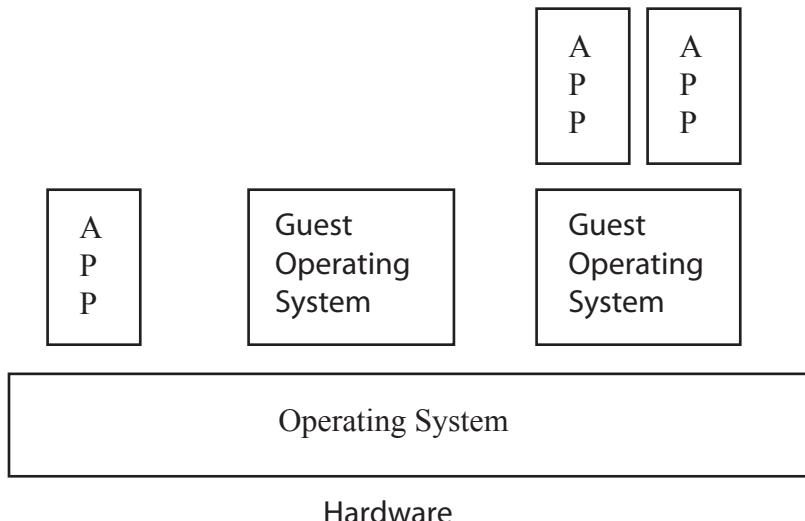


Figure 1.4: An operating system virtual machine

of physical memory, network bandwidth, and disk. Further, since the operating system must decide how to split the fixed set of resources among the various applications running at each moment, a particular application will have different amounts of resources from time to time, even when running on the same hardware. While a few applications might be designed to take advantage of a computer's specific hardware configuration and their specific resource assignment, most programmers want to use a higher level of abstraction.

We have just discussed one example of this: a uniprocessor can run only one program at a time, yet most operating systems allow multiple applications to appear to the user to be running at the same time. The operating system does so through a concept called *virtualization*. Virtualization provides an application with the illusion of resources that are not physically present. For example, the operating system can present to each application the abstraction that it has an entire processor dedicated to it, even though at a physical level there may be only a single processor shared among all the applications running on the computer. With the right hardware and operating system support, most physical resources can be virtualized: examples include the processor, memory, screen space, disk, and the network. Even the type of processor can be virtualized, to allow the same, unmodified application to be run on a smartphone, tablet, and laptop computer.

Pushing this a step further, some operating systems virtualize the entire computer, to run the operating system as an application running on top of another operating system (see Figure 1.4). This is called creating a *virtual machine*. The operating system running in the virtual machine, called the *guest operating system*, thinks it is running on a real, physical machine, but this is

Definition: **virtualization**

Definition: **virtual machine**

Definition: **guest operating system**

an illusion presented by the true operating system running underneath. One reason for the operating system to provide a virtual machine is for application portability. If a program only runs on an old version of an operating system, then we can still run the program on a new system running a virtual machine. The virtual machine hosts the application on the old operating system, running on top of the new operating system. Another reason for virtual machines is as an aid in debugging. If an operating system can be run as an application, then the operating system developers can set breakpoints, stop, and single step their code just as they would an application.

Definition: **atomic**

In addition to virtualization, operating systems mask many other limitations inherent in physical hardware, by providing applications with the illusion of hardware capabilities that are not physically present. For example, on a computer with multiple processors sharing memory, each processor can update only a single memory location at a time. The memory system in hardware ensures that any updates to the same memory word are *atomic*, that is, the value stored in memory is the last value stored by one of the processors, not a mixture of the updates of the different processors. Atomicity at the level of a memory word is preserved in hardware even if more than one processor attempts to write to memory at exactly the same time. While this might seem sufficient, applications (and the operating system itself) need to be able to update larger data structures, ones spread over many memory locations. What happens when two processors attempt to update the same data structure at roughly the same time? As we'll discuss later, the results can be quite unexpected and quite different from what would have happened had each of the processors updated the data structure in turn. Ideally, the programmer would like to have the abstraction of an atomic update to the entire data structure, not just to a single memory word. As we will discuss, the illusion of atomic updates to data structures is provided by the operating system using some specialized mechanisms provided in hardware.

Persistent block storage devices, such as magnetic disk or flash RAM, provide another example. At a physical level, these systems support block writes to storage, where the size of the block depends on physical device characteristics. If the computer crashes in the middle of a block write, it could leave the disk in an unknown state, with neither the old nor the new value stored at that location. Of course, applications need to be able to store data on disk that is variable in size, possibly spanning multiple disk blocks. And users want their data to be preserved even — or especially — if there is a machine failure while the disk is being updated.

We will discuss techniques that the operating system uses to accomplish these and other illusions. In each of these cases, the operating system provides a more convenient and flexible programming abstraction than what is provided by the underlying hardware.

Exercises

Take a moment to speculate; to build the systems we use today, someone needed to answer these questions. Consider how you might answer them, before seeing how others solved these puzzles.

4. How would you design combined hardware and software support to provide the illusion of a nearly infinite virtual memory on a limited amount of physical memory?
5. How would you design a system to run an entire operating system as an application running on top of another operating system?
6. How would you design a system to update complex data structures on disk in a consistent fashion despite machine crashes?

1.1.3 Common services: Operating system as glue

Operating system also play a third role: providing a set of common, standard services to applications to simplify and regularize their design. We saw an example of this with the web server outlined at the beginning of this chapter. The operating system hides the specifics of how the network and disk devices work, providing a simpler abstraction to applications based on receiving and sending reliable streams of bytes, and reading and writing named files. This allows the web server can focus on its core task of decoding incoming requests and filling them, rather than on the formatting of data into individual network packets and disk blocks.

An important reason for the operating system to provide common services, rather than leaving it up to each application, is to facilitate sharing between applications. The web server needs to be able to read the file that the text editor wrote. If applications are to share files, they need to be stored in a standard format, with a standard system for managing file directories. Likewise, most operating systems provide a standard way for applications to pass messages, and to share memory, to facilitate sharing.

The choice of which services an operating system should provide is often a matter of judgment. For example, computers can come configured with a blizzard of different devices: different graphics co-processors and pixel formats, different network interfaces (WiFi, Ethernet, and Bluetooth), different disk drives (SCSI, IDE), different device interfaces (USB, Firewire), and different sensors (GPS, accelerometers), not to mention different versions of each of those standards. Most applications will be able to ignore these differences, using only a generic interface provided by the operating system. For other applications,

such as a database, the specific disk drive may matter quite a bit. For those applications that can operate at a higher level of abstraction, the operating system serves as an interoperability layer, so that both applications, and the devices themselves, can be independently evolved without requiring simultaneous changes to the other side.

Another standard service in most modern operating systems is the graphical user interface library. Both Microsoft's and Apple's operating systems provide a set of standard user interface widgets. This facilitates a common "look and feel" to users, so that frequent operations such as pull down menus and "cut" and "paste" are handled consistently across applications.

Most of the code of an operating system is to implement these common services. However, much of the complexity of operating systems is due to resource sharing and masking hardware limits. Because the common service code is built on the abstractions provided by the other two operating system roles, this book will focus primarily on those two topics.

1.1.4 Operating system design patterns

The challenges that operating systems address are not unique — they apply to many different computer domains. Many complex software systems have multiple users, run programs written by third party developers, and/or need to coordinate many simultaneous activities. These pose questions of resource allocation, fault isolation, communication, abstractions of physical hardware, and how to provide a useful set of common services for software developers. Not only are the challenges the same, but often the solutions are as well: these systems use many of the design patterns and techniques described in this book.

For now, we focus on the challenges these systems have in common with operating systems:

- Cloud computing (Figure 1.5) is a model of computing where large-scale applications are run on shared computing and storage infrastructure in data centers, instead of on the user's own desktop computer. A similar approach is to run compute-intensive applications in the idle cycles of remote desktop computers. In both cases, many of the same issues arise as in operating systems, in terms of sharing, abstraction, and common services.
 - **Referee.** How are resources allocated between competing applications running in the cloud? How are buggy or malicious applications prevented from disrupting other applications?
 - **Illusionist.** The computing resources in the cloud are continually evolving; what abstractions are provided to isolate application developers from changes in the underlying hardware?

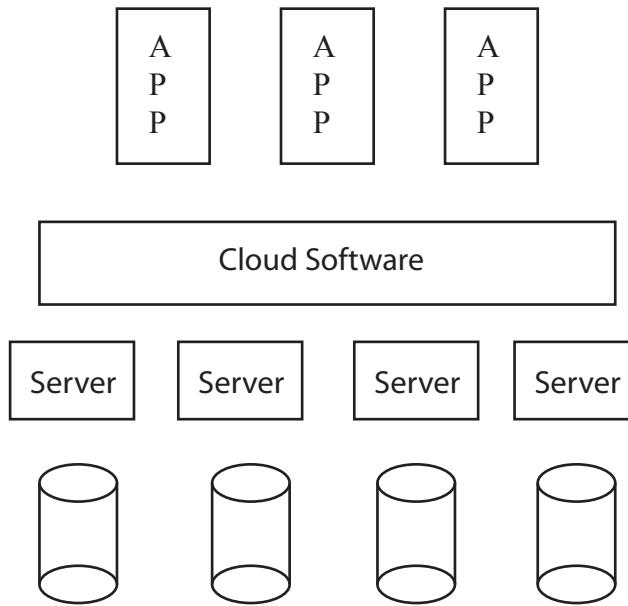
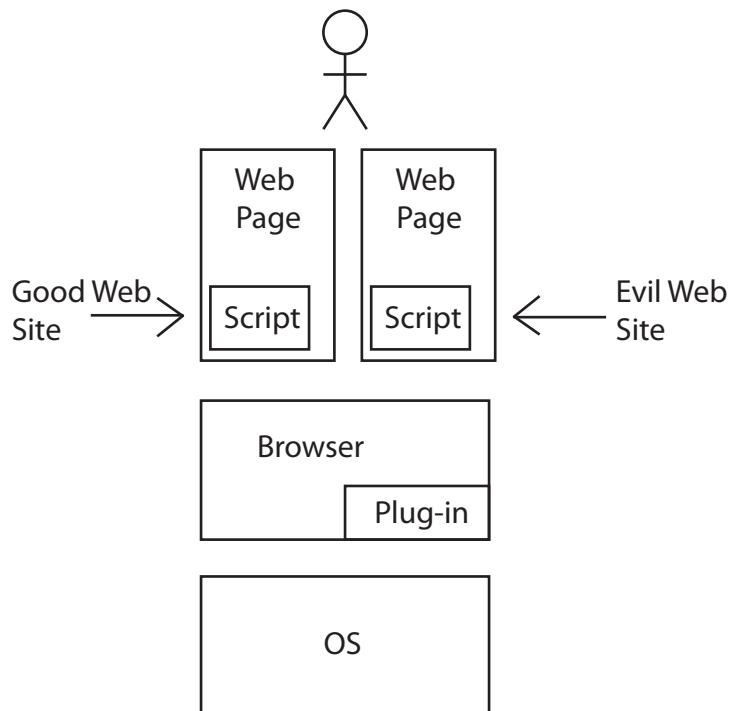


Figure 1.5: Cloud computing

- **Glue.** Cloud services often distribute their work across different machines. What abstractions should the cloud software provide to help services coordinate and share data between their various activities?
- Web browsers (Figure 1.6) such as Chrome, Internet Explorer, Firefox, and Safari each play a role similar to an operating system. Browsers load and display web pages, but as we mentioned earlier, many web pages embed scripting programs that the browser must execute. These scripts are often buggy and sometimes malicious; hackers have used them to take over vast numbers of home computers. Like an operating system, the browser must isolate the user, other web sites, and even the browser itself from errors or malicious activity by these scripts. Similarly, most browsers have a plug-in architecture for supporting extensions, and these extensions also need to be isolated from causing harm.
 - **Referee.** How can a browser ensure responsiveness, when a user has multiple tabs open and each tab is running a script from a different web site? How can we sandbox web scripts and plug-ins to prevent bugs from crashing the browser, and to prevent malicious scripts from accessing sensitive user data?
 - **Illusionist.** Many web services are geographically distributed for better fault tolerance. This way, if one server crashes or if its network connection has problems, the browser can connect to a different

**Figure 1.6:** Web browser

site. The user in most cases doesn't notice the difference, even when updating a shopping cart or web form. How does the browser mask server changes transparently to the user?

- **Glue.** How does the browser achieve a portable execution environment for scripts that works consistently across operating systems and hardware platforms?
- Media players, such as Flash and Silverlight, are often packaged as browser plug-ins, but they themselves provide an execution environment for scripting programs. Thus, these systems face many of the same issues as both the browsers and the operating systems on which they run: isolation of buggy or malicious code, concurrent background and foreground tasks, and plug-in architectures.
- Multi-user database systems (Figure 1.7) such as Oracle and Microsoft's SQL Server provide the ability for large organizations to store, query, and update large data sets, such as detailed records of every purchase ever made at Walmart. Large scale data analysis provides a huge benefit to optimizing business operations, but a consequence is that databases face many of the same challenges as operating systems. Databases are

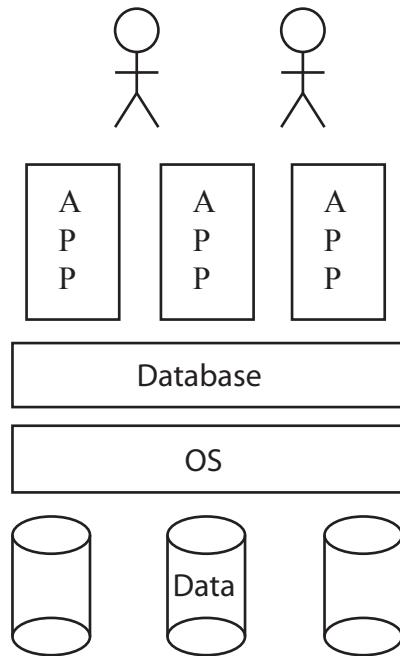
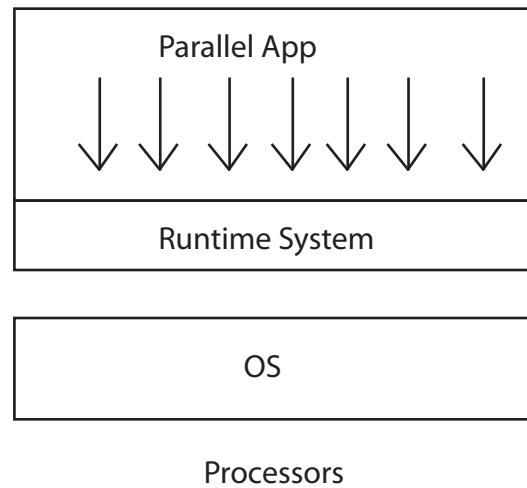
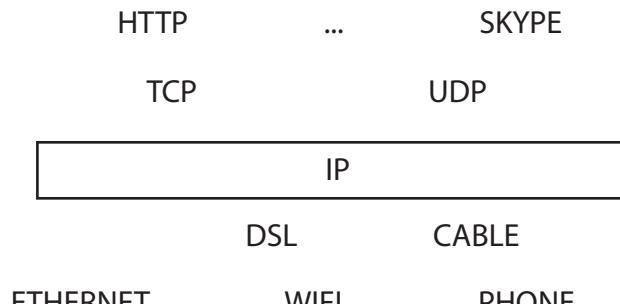


Figure 1.7: Database

simultaneously accessed by many different users in many different locations. Databases therefore need to allocate resources among different user requests, isolate concurrent updates to shared data, and ensure data is stored consistently on disk. In fact, several of the file system storage techniques we will discuss for operating systems, were originally developed for database systems.

- **Referee.** How should resources be allocated among the various users of a database? How does the database enforce data privacy so that only authorized users access relevant data?
- **Illusionist.** How does the database mask machine failures so that data is always stored consistently regardless of when the failure occurs?
- **Glue.** What common services make it easier for database users to develop their programs?
- Parallel applications (Figure 1.8) are programs that have been designed to take advantage of multiple processors on a single computer. Each application multiplexes its work onto a fixed number of processors and needs to ensure that accesses to shared data structures are coordinated to preserve consistency. While some parallel programs directly use the services pro-

**Figure 1.8:** Parallel Application**Figure 1.9:** Internet

vided by the underlying operating system, others need more careful control of the assignment of work to processors to achieve good performance. These systems interpose a runtime system on top of the operating system to manage user-level parallelism, essentially building a mini-operating system on top of the operating system.

- On the Internet (Figure 1.9), multiple users share the underlying physical network, posing the challenge of how the system should handle resource contention. The Internet is rife with malicious behavior, such as denial-of-service attacks that flood traffic on certain links to prevent legitimate users from communicating. Various attempts are underway to design solutions to allow the Internet to continue to function despite such attacks.
 - **Referee.** Should the Internet treat all users identically (e.g., network neutrality) or should ISPs have the ability to favor some uses over

others? Can the Internet be re-designed to prevent denial-of-service, spam, phishing, and other malicious behavior?

- **Illusionist.** The Internet provides the illusion of a single worldwide network, with the ability to deliver a packet from any machine on the Internet to any other machine. However, network hardware is in fact a large number of discrete network elements, with the ability to transmit limited size packets over a limited distance, and with some chance that the packet is garbled in the process. The Internet transforms the network into something more useful for applications like the web — a facility to reliably transmit data of arbitrary length, anywhere in the world.
- **Glue.** The Internet protocol suite was explicitly designed to act as an interoperability layer, to allow network applications to evolve independently of changes in network hardware, and vice versa. Does the success of the Internet hold any lessons for operating system design?

Many of these systems use the same techniques and design patterns as operating systems to address these challenges; studying operating systems is a great way to understand how these others systems work. In a few cases, different mechanisms are used to achieve the same goals, but even here, the boundary can be fuzzy. For example, browsers often use compile-time checks to prevent scripts from gaining control over the browser, while most operating systems use hardware-based protection to limit application programs from taking over the machine. More recently, however, some smartphone operating systems have begun to use the same compile-time techniques as browsers, but for protecting the smartphone operating system. In turn, some browsers have begun to use operating system hardware-based protection to improve the isolation they provide.

To avoid spreading our discussion too thinly, we focus this book on how operating systems work. Just as it is easier to learn a second computer programming language after you are fluent in the first, it is better to see how these operating systems principles are applied in one context before moving on to how these concepts are applied in other settings. We hope and expect however, that you will be able to apply the concepts in this book more widely than just operating system design.

Exercises

7. Society must also grapple with managing resources. What ways do we use for allocating resources, isolating misuse, and fostering sharing in real life?

1.2 Evaluation Criteria

Having defined what an operating system does, how should we choose among alternative approaches to the design challenges posed by operating systems? We next discuss several desirable criteria for operating systems. In many cases, tradeoffs between these criteria are inevitable — improving a system along one dimension will hurt it along another. We conclude with a discussion of some concrete examples of tradeoffs between these considerations.

1.2.1 Reliability

Perhaps the most important characteristic of an operating system is its reliability.

Definition: **Reliability**

Reliability is that a system does exactly what it is designed to do. As the lowest level of software running on the system, errors in operating system code can have devastating and hidden effects. If the operating system breaks, the user will often be unable to get any work done, and in some cases, may even lose previous work, e.g., if the failure corrupts files on disk. By contrast, application failures can be much more benign, precisely because operating systems provides fault isolation and a rapid and clean restart after an error.

Making the operating system reliable is challenging. Operating systems often operate in a hostile environment, where computer viruses and other malicious code may often be trying to take control of the system for their own purposes by exploiting design or implementation errors in the operating system's defenses.

Unfortunately, the most common ways for improving software reliability, such as running test cases for common code paths, are less effective when applied to operating systems. Since malicious attacks can target a specific vulnerability precisely to cause execution to follow a rare code path, literally everything has to work correctly for the operating system to be reliable. Even without malicious attacks that trigger bugs on purpose, extremely rare corner cases can occur regularly in the operating system context. If an operating system has a million users, a once in a billion event will eventually occur to someone.

A related concept is *availability*, the percentage of time that the system is usable. A buggy operating system that crashes frequently, losing the user's work, is both unreliable and unavailable. A buggy operating system that crashes frequently, but never loses the user's work and cannot be subverted by a malicious attack, would be reliable but unavailable. An operating system that has been subverted, but continues to appear to run normally while logging the user's keystrokes, is unreliable but available.

Thus, both reliability and availability are desirable. Availability is affected by two factors: the frequency of failures, called the *mean time to failure (MTTF)*, and the time it takes to restore a system to a working state after a failure (for example, to reboot), the *mean time to repair (MTTR)*. Availability can be im-

Definition: **mean time to failure (MTTF)**

Definition: **mean time to repair (MTTR)**

proved by increasing the MTTF or reducing the MTTR, and we will present operating systems techniques that do each.

Throughout this book, we will present various approaches to improving operating system reliability and availability. In many cases, the abstractions may seem at first glance overly rigid and formulaic. It is important to realize this is done on purpose! Only precise abstractions provide a basis for constructing reliable and available systems.

Exercises

8. Suppose you were tasked with designing and implementing an ultra-reliable and ultra-available operating system. What techniques would you use? What tests, if any, might be sufficient to convince you of the system's reliability, short of handing your operating system to millions of users to serve as beta testers?
9. MTTR, and therefore availability, can be improved by reducing the time to reboot a system after a failure. What techniques might you use to speed up booting? Would your techniques always work after a failure?

1.2.2 Security

Two concepts closely related to reliability are security and privacy. *Security* is the property that the computer's operation cannot be compromised by a malicious attacker. *Privacy* is a part of security — that data stored on the computer is only accessible to authorized users.

Alas, no useful computer is perfectly secure! Any complex piece of software has bugs, and even otherwise innocuous bugs can be exploited by an attacker to gain control of the system. Or the hardware of the computer might be tampered with, to provide access to the attacker. Or the computer's administrator might turn out to be untrustworthy, using their privileges to steal user data. Or the software developer of the operating system might be untrustworthy, inserting a backdoor for the attacker to gain access to the system.

Nevertheless, an operating system can, and should, be designed to minimize its vulnerability to attack. For example, strong fault isolation can prevent third party applications from taking over the system. Downloading and installing a screen saver or other application should not provide a way for a malicious attacker to surreptitiously install a *computer virus* on the system. A computer virus is a computer program that modifies an operating system or application to provide the attacker, rather than the user, control over the system's resources

or data. An example computer virus is a keylogger: a program that modifies the operating system to record every keystroke entered by the user and send those keystrokes back to the attacker's machine. In this way, the attacker could gain access to the user's passwords, bank account numbers, and other private information. Likewise, a malicious screen saver might surreptitiously scan the disk for files containing personal information or turn the system into an email spam server.

Even with strong fault isolation, a system can be insecure if its applications are not designed for security. For example, the Internet email standard provides no strong assurance of the sender's identity; it is possible to form an email message with anyone's email address in the "from" field, not necessarily the actual sender. Thus, an email message can appear to be from someone (perhaps someone you trust), when in reality it is from someone else (and contains a malicious virus that takes over the computer when the attachment is opened). By now, you are hopefully suspicious of clicking on any attachment in an email. If we step back, though, the issue could instead be cast as a limitation of the interaction between the email system and the operating system — if the operating system provided a cheap and easy way to process an attachment in an isolated execution environment with limited capabilities, then even if the attachment contained a virus, it would be guaranteed not to cause a problem.

Complicating matters is that the operating system must not only prevent unwanted access to shared data, it must also *allow* access in many cases. We want users and programs to interact with each other, to be able to cut and paste text between different applications, and to read or write data to disk or over the network. If each program was completely standalone, and never needed to interact with any other program, then fault isolation by itself would be enough. However, we not only want to be able to isolate programs from one another, we also want to be able to easily share data between programs and between users.

Thus, an operating system needs both an enforcement mechanism and a security policy. *Enforcement* is how the operating system ensures that only permitted actions are allowed. The *security policy* defines what is permitted — who is allowed to access what data and who can perform what operations. Malicious attackers can target vulnerabilities in either enforcement mechanisms or security policy.

1.2.3 Portability

All operating systems provide applications an abstraction of the underlying computer hardware; a *portable* abstraction is one that does not change as the hardware changes. A program written for Microsoft's Windows 7 should run correctly regardless of whether a specific graphics card is being used, whether persistent storage is provided via flash memory or rotating magnetic disk, or whether the network is Bluetooth, WiFi, or gigabit Ethernet.

Portability also applies to the operating system itself. Operating systems are among the most complex software systems ever invented, so it is impractical to re-write them from scratch every time some new hardware is produced or every time a new application is developed. Instead, new operating systems are often derived, at least in part, from old ones. As one example, iOS, the operating system for the iPhone and iPad, is derived from the OS X code base.

As a result, most successful operating systems have a lifetime measured in decades: the initial implementation of Microsoft Windows 8 began with the development of Windows NT starting in 1990, when the typical computer was more than 10000 times less powerful and had 10000 times less memory and disk storage, than is the case today. Operating systems that last decades are no anomaly: Microsoft's prior operating system code base, MS/DOS, was first introduced in 1981. It later evolved into the early versions of Microsoft Windows before finally being phased out around 2000.

This means that operating systems need to be designed to support applications that have not been written yet and to run on hardware that has yet to be developed. Likewise, we do not want to have to re-write applications as the operating system is ported from machine to machine. Sometimes of course, the importance of “future-proofing” the operating system is discovered only in retrospect. Microsoft’s first operating system, MS/DOS, was designed in 1981 assuming that personal computers had no more than 640KB of memory. This limitation was acceptable at the time, but today, even a cellphone has orders of magnitude more memory than that.

How might we design an operating system to achieve portability? We will discuss this in more depth, but an overview is provided above in Figure 1.3. For portability, it helps to have a simple, standard way for applications to interact with the operating system, through the abstract machine interface. The *abstract machine interface (AMI)* is the interface provided by operating systems to applications. A key part of the AMI is the *application programming interface (API)*, the list of function calls the operating system provides to applications. The AMI also includes the memory access model and which instructions can be legally executed. For example, an instruction to change whether the hardware is executing trusted operating system code, or untrusted application code, needs to be available to the operating system but not to applications.

Definition: **abstract machine interface (AMI)**

Definition: **application programming interface (API)**

A well-designed operating system AMI provides a fixed point across which both application code and hardware can evolve independently. This is similar to the role of the Internet Protocol (IP) standard in networking — distributed applications such as email and the web, written using IP, are insulated from changes in the underlying network technology (Ethernet, WiFi, optical). Equally important is that changes in applications, from email to instant messaging to file sharing, do not require simultaneous changes in the underlying hardware.

This notion of a portable hardware abstraction is so powerful that operating

Definition: **hardware abstraction layer (HAL)**

systems use the same idea *internally*, so that the operating system itself can largely be implemented independently of the specifics of the hardware. This interface is called the *hardware abstraction layer (HAL)*. It might seem at first glance that the operating system AMI and the operating system HAL should be identical, or nearly so — after all, both are portable layers designed to hide unimportant hardware details. The AMI needs to do more, however. As we noted, applications execute in a restricted, virtualized context and with access to high level common services, while the operating system itself is implemented using a procedural abstraction much closer to the actual hardware.

Today, Linux is an example of a highly portable operating system. Linux has been used as the operating system for web servers, personal computers, tablets, netbooks, ebook readers, smartphones, set top boxes, routers, WiFi access points, and game consoles. Linux is based on an operating system called UNIX, originally developed in the early 1970's. UNIX was written by a small team of developers, and because they could not afford to write very much code, it was designed to be very small, simple to program against, and highly portable, at some cost in performance. Over the years, UNIX's and Linux's portability and convenient programming abstractions have been keys to its success.

1.2.4 Performance

While the portability of an operating system can become apparent over time, the performance of an operating system is often immediately visible to its users. Although we often associate performance with each individual application, the operating system's design can have a large impact on the application's perceived performance because it is the operating system that decides when an application can run, how much memory it can use, and whether its files are cached in memory or clustered efficiently on disk. The operating system also mediates application access to memory, the network, and the disk. The operating system needs to avoid slowing down the critical path while still providing needed fault isolation and resource sharing between applications.

Performance is not a single quantity, but rather it can be measured in several different ways. One performance metric is the *efficiency* of the abstraction presented to applications. A related concept to efficiency is *overhead*, the added resource cost of implementing an abstraction. One way to measure efficiency (or inversely, overhead) is the degree to which the abstraction impedes application performance. Suppose the application were designed to run directly on the underlying hardware, without the overhead of the operating system abstraction; how much would that improve the application's performance?

Operating systems also need to allocate resources between applications, and this can affect the performance of the system as perceived by the end user. One issue is *fairness*, between different users of the same machine, or between different applications running on that machine. Should resources be divided

equally between different users or different applications, or should some get preferential treatment? If so, how does the operating system decide what tasks get priority?

Two related concepts are *response time* and *throughput*. Response time, sometimes called *delay*, is how long it takes for a single specific task from when it starts until it completes. For example, a highly visible response time for desktop computers is the time from when the user moves the hardware mouse until the pointer on the screen reflects the user's action. An operating system that provides poor response time can be unusable. Throughput is the rate at which a group of tasks can be completed. Throughput is a measure of efficiency for a group of tasks rather than a single one. While it might seem that designs that improve response time would also necessarily improve throughput, this is not the case, as we will discuss later in this book.

A related consideration is performance *predictability*, whether the system's response time or other metric is consistent over time. Predictability can often be more important than average performance. If a user operation sometimes takes an instant, and sometimes much longer, the user may find it difficult to adapt. Consider, for example, two systems. In one, the user's keystrokes are almost always instantaneous, but 1% of the time, a keystroke takes 10 seconds to take effect. In the other system, the user's keystrokes always take 0.1 seconds to be reflected on the screen. Average response time may be the same in both systems, but the second is more predictable. Which do you think would be more user-friendly?

For a simple example illustrating the concepts of efficiency, overhead, fairness, response time, throughput, and predictability, consider a car driving to its destination. If there were never any other cars or pedestrians on the road, the car could go quite quickly, never needing to slow down for stop lights. Stop signs and stop lights enable cars to share the road, at some cost in overhead and response time for each individual driver. As the system becomes more congested, predictability suffers. Throughput of the system is improved with carpools. In congested situations and especially with dedicated carpool lanes, carpools can also improve latency even though carpools need to coordinate their pick-ups. Predictability, throughput, and arguably fairness can all be improved by scrapping the car and installing mass transit.

1.2.5 Adoption

In addition to reliability, portability and performance, the success of an operating system depends on two factors outside its immediate control: the (wide) availability of applications ported to that operating system, and the (wide) availability of hardware that the operating system can support. An iPhone runs iOS, but without the preinstalled applications and the contents of the App Store, the iPhone would be just a cellphone with (allegedly) bad phone reception.

Definition: **response time**

Definition: **throughput**

Definition: **predictability**

Definition: **network effect**

The *network effect* occurs when the value of some technology depends not only on its intrinsic capabilities, but also on the number of other people who have adopted that technology. Application and hardware designers spend their efforts on those operating system platforms with the most users, while users favor those operating systems with the best applications or the cheapest hardware. If this sounds circular, it is! More users imply more applications and cheaper hardware; more applications and cheaper hardware imply more users, in a virtuous cycle.

Consider how you might design an operating system to take advantage of the network effect, or at least to avoid being crushed by it. An obvious step would be designing the system to make it easy to accommodate new hardware, and to make it easy for applications to be ported across different versions of the same operating system.

A more subtle issue is the choice of whether the operating system programming interface (API), or the operating system source code itself, is open or

Definition: **proprietary** proprietary. A *proprietary* system is one under the control of a single company, so it can be changed at any time by its provider to meet the needs of

Definition: **open system** its customers. An *open system* is one where the system's source code is public, allowing anyone the ability to inspect the code and change it. Often, an open system will have an API that can only be changed with the agreement of a public standards body. Adherence to standards provides assurance to the application developer that the API will not be changed except by general agreement; on the other hand, standards bodies can make it difficult to quickly add new, desired features.

Neither open systems nor proprietary ones are obviously better for widespread adoption. Windows 7 and MacOS are examples of proprietary operating systems; Linux is an example of an open operating system. All three are widely used! Open systems are easier to adapt to a wide variety of hardware platforms, but risk fragmentation, impairing the network effect. Purveyors of proprietary operating systems argue that their systems are more reliable and better adapted to the needs of their customers. Interoperability problems are reduced if both the hardware and software are controlled by the same company, but limiting an operating system to one hardware platform impairs the network effect.

Making it easy to port applications from existing systems to a new operating system can help a new system become established, and conversely, designing an operating system API to make it difficult to port applications away from the operating system can help prevent competition from becoming established. Thus, there are often commercial pressures for operating system interfaces to become idiosyncratic. Throughout this book, we will discuss operating systems issues at a conceptual level, but it is important to realize that the details will vary quite a bit for any specific operating system, due to important, but also somewhat chaotic, commercial interests.

Android vs. iPhone

One avenue to improving system reliability might be to limit third party applications, or to vet them in some way. Of course, limiting applications can hurt adoption. Two operating systems vendors taking opposite positions on this recently are Apple and Google. For the iPhone, Apple requires pre-approval before any application can be loaded on the iPhone, possibly enhancing reliability. In practice, however, it can be difficult to verify all aspects of application behavior, e.g., to prevent a game application from downloading telephone numbers stored on the smartphone for telemarketing purposes. Thus, it is unclear how much benefit users will have in practice. Google takes the opposite approach: it gives the users control over which applications can be installed on Android phones, possibly enhancing wider user adoption, but potentially hurting system reliability. It will be interesting to see which approach is more successful.

1.2.6 Tradeoffs

Most practical operating system designs need to strike a balance between the goals of reliability, security, portability, performance, and adoption. Design choices that improve portability — for example, by preserving legacy interfaces — often make the system as a whole less reliable and less secure. Similarly, there will often be ample room for breaking an abstraction to tweak some added performance out of the system. However, such performance optimizations come at a cost of added complexity and therefore potentially decreased reliability. The operating system designer must carefully weigh these competing goals.

To illustrate the tradeoff between performance and complexity, we relate the following true story. An operating system was designed and implemented in the late 1980's, using a type-safe language to reduce the incidence of programmer errors. For speed, the most frequently used routines at the core of the operating system were implemented in assembly code. In one of these routines, the implementers decided to use a sequence of instructions that shaved a single instruction off a very frequently used code path, but that would sometimes break if the operating system exceeded a particular size. At the time, the operating system was nowhere near this limit. After a few years of production use, however, the system started mysteriously crashing, apparently at random, and only after many days of execution. Many weeks of painstaking investigation revealed the problem: the operating system had grown beyond the limit assumed in the assembly code implementation. The fix was easy, once the problem was found, but the question for the reader is: do you think the original optimization was worth the risk?

Exercises

	1981	1996	2011	factor
MIPS	1	300	10000	10K
MIPS/\$	\$100K	\$30	\$0.50	200K
DRAM	128KB	128MB	10GB	100K
Disk	10MB	4GB	1TB	100K
Home Internet	9.6 Kbps	256 Kbps	5 Mbps	500
LAN network	3 Mbps (shared)	10 Mbps	1 Gbps	300
Users per machine	100	1	<< 1	100+

Figure 1.10: Computer performance over time

- For the computer you are currently using, how should the operating system designers prioritize among reliability, security, portability, performance, and adoption? Explain why.

1.3 A brief history of operating systems

We conclude this chapter with a discussion of the origins of operating systems, as a way of illustrating where operating systems are headed in the future. As the lowest layer of software running on top of computer hardware, operating systems have been around nearly as long as the first computers, and they have evolved nearly as rapidly as computer hardware.

1.3.1 Impact of technology trends on operating systems

The most striking aspect of the last fifty years in computing technology has been the cumulative effect of Moore's Law, and the comparable advances in related technologies such as memory and disk storage. Moore's Law states that transistor density increases exponentially over time; similar exponential improvements have occurred in many other component technologies. Figure 1.10 provides an overview of the past thirty years of technology improvements in computer hardware. The cost of processing has decreased by over five orders of magnitude over the past thirty years; the cost of memory and disk capacity has followed a similar trajectory. Of course, not all technologies have improved at the same rate; disk latency has improved over time, but at a much slower rate

than disk capacity. These relative changes have radically altered both the use of computers and the tradeoffs faced by the operating system designer.

It is hard to imagine how things used to be. Today, we are able to carry smartphones with incredibly powerful computers around in our pockets. Thousands of server computers wait patiently for a user to type in a search query; when the query arrives, the servers can synthesize a response in a fraction of a second. In the early years of computing, however, the computers were more expensive than the salaries of the people who used them. Users would queue up, often for days, for their turn to run a program. A similar progression from expensive to cheap devices occurred with telephones over the past hundred years. Initially, telephone lines were very expensive, so that a single line was shared among everyone in a neighborhood. Over time, of course, both computers and telephones have become cheap enough to sit idle until we need them.

Despite these changes, operating systems still face the same conceptual challenges as they did fifty years ago. To manage computer resources for applications and users, operating systems must allocate resources among applications, provide fault isolation and communication services, abstract hardware limitations, and so forth. Tremendous progress has been made towards improving the reliability, security, efficiency, and portability of operating systems, but much further progress is still needed. Despite the fact we do not know how computing technology or application demand will evolve over the next 10-20 years, it is highly likely we will continue to need to address these fundamental operating system challenges in the future.

1.3.2 Early operating systems

The first operating systems were runtime libraries, intended to simplify programming early computer systems. Rather than the tiny, inexpensive yet massively complex hardware and software systems we have today, the first computers often took up an entire floor of a warehouse, cost millions of dollars, yet were only capable of being used by a single person at a time. The user would first reset the computer, load in their program, and hit go, producing output that could be pored over while the next user took their turn. If the user made an error, they needed to wait their turn to try the run over again, often the next day.

Although it might seem like there was no need for an operating system in this setting, if computers are enormously expensive, anything that reduces the likelihood of programmer error is extremely valuable. The first operating systems were seen as a way of reducing errors by providing a standard set of common services. For example, early operating systems provided standard routines to perform input/output (I/O) processing, which each user could link into their program. By using these services, a user's program would be more likely to run correctly and produce useful output.

Even though these initial operating systems were a huge step forward, the

result was still horribly inefficient. It was around this time that the CEO of IBM famously predicted that there would only ever be a market for five computers in the world. If computers still cost millions of dollars and could only be used to run the tiny applications of the time by a single person at a time, he would probably have been proven right!

1.3.3 Multi-user operating systems

The next step forward was sharing, introducing many the advantages, and challenges, that we see from operating systems today. If processor time is incredibly valuable, restricting the system to one user at a time is incredibly wasteful. The processor is idle while the program is being loaded and as the data in being input or the results being output, even though there was usually a long line of people waiting their turn to use the processor.

Definition: **batch operating systems**

With *batch operating systems*, one program can be using the processor while another is being loaded into memory. The batch operating system was installed in the system's memory, and ran a simple loop: load, run, and unload each job in turn. While one job was running, the operating system would set up the I/O devices to do background transfers for the next/previous job using a process called *direct memory access (DMA)*. With DMA, the I/O device transfers its data directly into memory at a location specified by the operating system. When the I/O transfer completes, the hardware interrupts the processor, transferring control to the operating system interrupt handler. The operating system starts the next DMA transfer and then resumes execution of the application. The interrupt appears to the application as if nothing had happened, except for some delay between one instruction and the next.

Definition: **direct memory access (DMA)**

Batch operating systems were soon extended to run multiple applications at once, that is, *multitasking*, or what is also sometimes called *multiprogramming*.

Definition: **multiprogramming**

Multiple programs were loaded into memory at the same time, each ready to use the processor if for any reason the previous task needed to pause, for example, to read some additional input or produce some output. Multitasking improves processor efficiency essentially to 100%; provided the queue of tasks is long enough, and there are a sufficient number of I/O devices to keep feeding the processor, there is never a need for the processor to wait for work.

However, processor sharing raises the problem that programs need to be isolated from one another, if only to protect against a bug in one program crashing or corrupting another. During this period, computer designers added hardware memory protection, as a way of reducing the overhead of fault isolation.

A practical challenge with batch computing, however, was how to debug the operating system itself. Unlike an application program, a batch operating system assumes it is in direct control of the hardware. New versions could only be tested by stopping every application and rebooting the system, essentially turning the computer back into a single user system. Needless to say, this was

an expensive operation, and so it typically was scheduled for the dead of the night.

Virtual machines were developed to address this limitation (see Figure 1.4). Instead of running a test operating system directly on the hardware, with virtual machines an operating system can be run just like any other application. The true operating system, called a *virtual machine monitor*, exports an abstract machine interface (AMI) that is identical to the underlying hardware. The test operating system running on top of the virtual machine does not need to know that it is running in a virtual environment — it executes instructions, accesses hardware devices, restores application state after an interrupt just as it would running on real hardware.

Definition: **virtual machine monitor**

Virtual machines have become widely used for operating system development, backward compatibility, and cross-platform support. Old application software that runs only on an old version of an operating system can share hardware with entirely new applications. The virtual machine monitor runs two virtual machines — one for the new operating system for up to date applications and a separate one for any legacy applications. As another example, MacOS users wanting to run Windows applications can do so by running them inside a virtual machine.

1.3.4 Time-sharing operating systems

Eventually, the cumulative effect of Moore’s Law meant that the cost of computing dropped to where we could start designing systems optimized for users, rather than optimized for efficient use of the processor. UNIX, for example, was developed in the early 70’s on a spare computer that no one was using at the time. UNIX became the basis of Apple’s MacOS X, Linux, widely used for servers at Google and Facebook, VMware, a widely used virtual machine monitor, and Google Android, a smartphone operating system. Figure 1.11 provides a sketch of the history of some of today’s commercial operating systems.

A *time-sharing* operating system, such as Windows, MacOS, and Linux, is one designed to support interactive use of the computer, rather than the batch mode processing of earlier systems. With time-sharing, the user types input on a keyboard or other input device directly connected to the computer. Each keystroke or mouse action causes an interrupt to the processor signalling the event; the interrupt handler reads the event from the device and queues it inside the operating system. When the user’s word processor, game, or other application resumes, it fetches the event from the operating system, processes the event, and alters the display appropriately, before fetching the next event. Hundreds or even thousands of such events can be processed per second, requiring both the operating system and the application to be designed for frequent, very short bursts of activity, rather than the sustained execution model of batch processing.

Definition: **time-sharing**

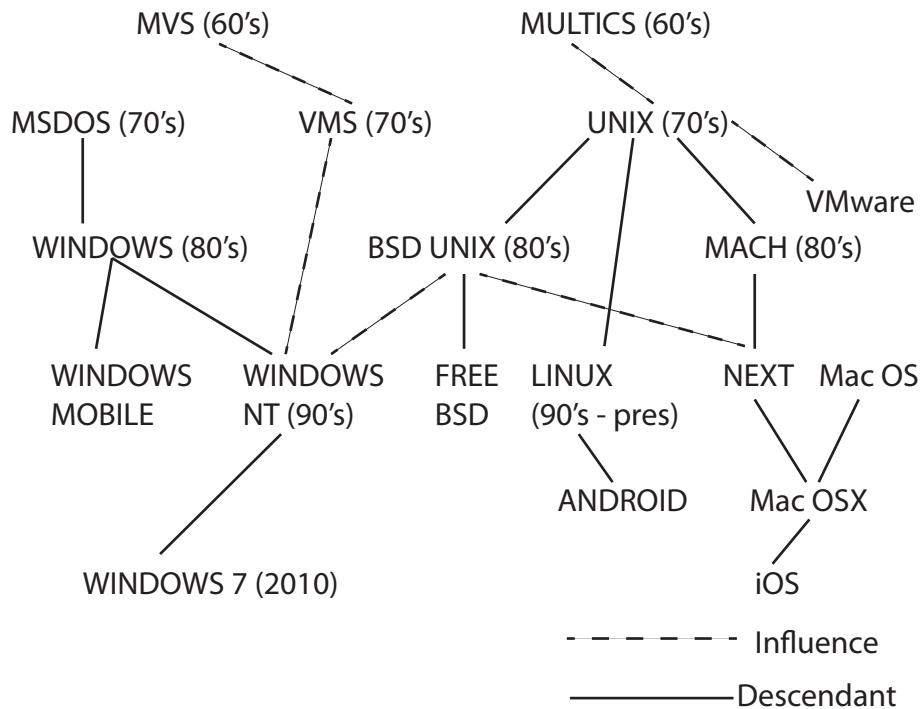


Figure 1.11: Genealogy of several modern operating systems.

The basic operation of a web server is similar to a time-sharing system. The web server waits for a packet to arrive, requesting a web page or to perform an action such as search the web or purchase a book. The network hardware copies the arriving packet into memory using DMA. Once the transfer is complete, it signals the arrival of a packet by interrupting the processor, triggering the server to perform the requested task. Likewise, the processor is interrupted as each block of the web page is read from disk into memory. Like a time-sharing system, server operating systems need to be designed to handle very large numbers of short actions per second.

The earliest time-sharing systems supported many simultaneous users, but even this was only a phase. Eventually, computers became cheap enough that people could afford their own dedicated "personal" computers, which would sit patiently unused for much of the day. Instead, access to shared data became paramount, cementing the shift to client-server computing.

1.3.5 Modern operating systems

Today, we have vast diversity of computing devices, and many different operating systems running on those devices. The tradeoffs faced by an operating

system designer depend on the physical capabilities of the hardware, and on the application and user demand for that hardware. Here are some examples of operating systems that the reader might have used recently:

- **Desktop, laptop, and netbook operating systems.** Examples include Windows 7, MacOS, and Linux. These systems are characterized by having a single user at a time, many applications, and many I/O devices. One might think that with only one user, there is no need to design the system to support sharing, and indeed the initial personal computer operating systems took this approach. They had very limited ability to isolate different parts of the system from one another. Over time, however, it became clear that stricter fault isolation was needed to improve system reliability and resilience against computer viruses.
- **Smartphone operating systems.** A smartphone is a cellphone with an embedded computer capable of running third party applications. Examples of smartphone operating systems include iOS, Android, Symbian, WebOS, Blackberry OS and Windows Phone. Obviously smartphones have only one user, but they still must support many applications. Because applications downloaded off the Internet might have viruses that attempt to surreptitiously take control over the phone, the operating system must be designed to protect itself from misbehaving applications and to protect applications from each other.
- **Embedded systems.** Over time, computers have become cheap enough to integrate into any number of consumer devices, from cable TV set top boxes, to microwave ovens, to the control systems for automobiles and airplanes, to LEGO robots, and to medical devices such as MRI machines and WiFi based intravenous titration systems. Embedded devices typically run a customized operating system bundled with the task-specific software to control the device. Although you might think these systems as too simple to merit much attention, software errors can have quite devastating effects. One of the earliest documented examples of a computer causing human deaths was the Therac-25 in the mid-1980's. The problem occurred because of programming errors in the operating system code for a medical device, errors that could have been prevented had the system developers followed the design paradigms outlined in this book.
- **Virtual machines.** As we mentioned, a virtual machine monitor is an operating system that can run another operating system as if it were an application. Example virtual machine monitors include VMWare, Xen, and Windows Virtual PC. Virtual machine monitors face many of the same challenges as other operating systems, with the added challenge posed by coordinating a set of coordinators. The operating system running inside a virtual machine makes resource allocation and fault isolation decisions as if it is in complete control of its resources, even though it is in fact sharing the system with other operating systems and applications.

- **Server operating systems.** Search engines, web media, e-commerce sites, and email systems are hosted on computers in data centers; each of these computers runs an operating system, often an industrial strength version of one of the desktop systems described above. Usually, only a single application, such as a web server, runs per machine, but the operating system needs to coordinate thousands of simultaneous incoming network connections onto that application. Servers operate in a hostile environment, sometimes receiving many thousands of packets per second attempting to subvert or block the service. At the same time, there is a premium on responsiveness. Amazon and Google both report that adding even a sub-second delay to each web request can dramatically reduce their revenue per user.
- **Server clusters.** For fault tolerance and scale, many web sites are implemented on distributed clusters of computers housed in one or more geographically distributed data centers. If one computer fails due to a hardware fault, software crash, or power failure, another computer can take over its role. If demand for the web site exceeds what a single computer can accommodate, web requests can be partitioned among multiple machines. As with normal operating systems, server cluster applications run on top of an abstract cluster interface to isolate the application from hardware changes and to isolate faults in one application from affecting other applications in the same data center. Likewise, resources can be shared between various applications on the same web site (such as Google search, Google earth, and gmail), and resources can be shared between multiple web sites hosted on the same cluster hardware (such as with Amazon's Elastic Compute Cloud).

1.3.6 Future operating systems

Where do operating systems go from here over the next decade? Operating systems have become dramatically better at resisting malicious attacks over the past decade, but they still have quite a ways to go. Provided security and reliability challenges can be met, there are huge potential benefits from having computers tightly control and coordinate physical infrastructure such as the power grid, the telephone network, and a hospital's medical devices and medical record systems. Thousands of lives are lost annually through traffic accidents that could potentially be prevented through computer control of automobiles. If we are to rely on computers for these critical systems, we need greater assurance that operating systems are up to the task.

Beyond mission critical systems, whenever the underlying hardware changes, there is work to do for the operating system designer. The future of operating systems is also the future of hardware:

- Very large scale data centers, coordinating hundreds of thousands or even

millions of computers to support some essential online service.

- Very large scale multicore systems. Computer architectures have already switched to providing multiple processors per chip; this trend will continue, potentially yielding systems with hundreds or possibly even thousands of processors per machine.
- Ubiquitous portable computing devices, including smartphones, tablets, and ebook readers. Computers are likely to become untethered from the keyboard and the screen, responding to voice, gestures, and perhaps even brain waves.
- Very heterogeneous systems, as every device becomes programmable, from supercomputers to refrigerators down to individual light switches.
- Very large scale storage: everything that can be stored, will be, and needs to be stored reliably so that it can be retrieved at any point, even decades later.

Managing all this is the job of the operating system.

Exercises

For convenience, the exercises from the body of the chapter are repeated here.

1. Suppose a computer system and all of its applications are completely bug free. Suppose further that everyone in the world is completely honest and trustworthy. In other words, we do not need to consider fault isolation.
 - a. How should the operating system allocate time on the processor? Should it give all of the processor to each application until it no longer needs it? If there are multiple tasks ready to go at the same time, should it schedule the task with the least amount of work to do or the one with the most? Justify your answer.
 - b. How should the operating system allocate physical memory between applications? What should happen if the set of applications do not all fit in memory at the same time?
 - c. How should the operating system allocate its disk space? Should the first user to ask be able to grab all of the free space? What would the likely outcome be for that policy?
2. Now suppose the computer system needs to support fault isolation. What hardware and/or operating support do you think would be needed to accomplish this goal?
 - a. For protecting an application's data structures in memory from being corrupted by other applications?
 - b. For protecting one user's disk files from being accessed or corrupted by another user?
 - c. For protecting the network from a virus trying to use your computer to send spam?
3. How should an operating system support communication between applications?
 - a. Through the file system?
 - b. Through messages passed between applications?
 - c. Through regions of memory shared between the applications?
 - d. All of the above? None of the above?
4. How would you design combined hardware and software support to provide the illusion of a nearly infinite virtual memory on a limited amount of physical memory?

5. How would you design a system to run an entire operating system as an application running on top of another operating system?
6. How would you design a system to update complex data structures on disk in a consistent fashion despite machine crashes?
7. Society must also grapple with managing resources. What ways do we use for allocating resources, isolating misuse, and fostering sharing in real life?
8. Suppose you were tasked with designing and implementing an ultra-reliable and ultra-available operating system. What techniques would you use? What tests, if any, might be sufficient to convince you of the system's reliability, short of handing your operating system to millions of users to serve as beta testers?
9. MTTR, and therefore availability, can be improved by reducing the time to reboot a system after a failure. What techniques might you use to speed up booting? Would your techniques always work after a failure?
10. For the computer you are currently using, how should the operating system designers prioritize among reliability, security, portability, performance, and adoption? Explain why.

Part I

Kernels and Processes

Chapter 2

The Kernel Abstraction

Strong fences make good neighbors.

– 17th century proverb

A central role of operating systems is *protection* — the isolation of potentially misbehaving applications and users so that they do not corrupt other applications or the operating system itself. Protection is essential to achieving several of the goals we listed for operating systems in the previous chapter:

Definition: **protection**

- **Reliability.** Protection is needed to prevent bugs in one program from causing crashes in other programs or in the operating system. To the user, a system crash will appear to be the operating system's fault, even if the root cause of the problem was some unexpected behavior by an application or user. Thus, for high system reliability, an operating system must bullet proof itself so that it operates correctly regardless of whatever an application or user might do.
- **Security.** Some users or applications on a system may be less than completely trustworthy and therefore the operating system needs to limit the scope of what they can do. Without protection, a malicious user might surreptitiously change application files or even the operating system itself, leaving the user none the wiser. For example, if a malicious application is permitted to write directly to the disk, it could modify the file containing the operating system's code, so that the next time the system starts, the

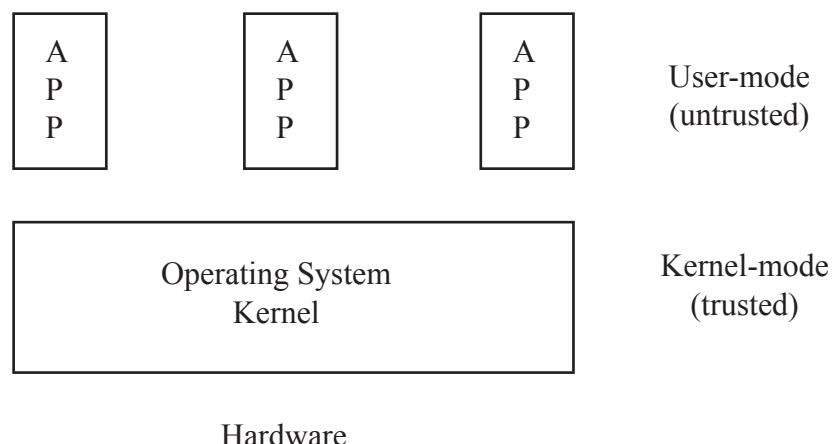


Figure 2.1: User-mode and kernel-mode operation.

modified operating system will boot instead, installing spyware and disabling virus protection. For security, an operating system must prevent untrusted code from modifying system state.

- **Privacy.** On a multi-user system, each user must be limited to just the data that she is permitted to access. Without protection provided by the operating system, any user or application running on a system could access any of the data stored on the system, without the knowledge or approval of the data's owner. For example, hackers often use popular applications such as screen savers as a way to gain access to personal emails, telephone numbers, and credit card data stored on the system. For privacy, an operating system must prevent untrusted code from accessing unauthorized data.
- **Efficiency.** Protection is also needed for effective resource allocation. Without protection, an application can gather any amount of processing time, memory, or disk space that it wants. On a single-user system, this means that a buggy application can prevent other applications from running, or simply make them run so slowly that they appear to be stalled. On a multi-user system, one user could grab all of the system's resources for herself. Thus, for efficiency and fairness, an operating system must be able to limit the amount of resources assigned to each application or user.

Definition: **operating system kernel**

Implementing protection is the job of the *operating system kernel*. The kernel is the lowest level of software running on the system, with full access to all of the capabilities of the hardware. The kernel is necessarily *trusted* to do anything that can be done with the hardware. Everything else — that is, the untrusted software running on the system — is run in a restricted environment,

with less than complete access to the full power of the hardware. Figure 2.1 illustrates this difference between kernel-level and user-level execution.

In turn, applications themselves often need to safely execute untrusted third party code. An example is a web browser executing embedded Javascript to draw a web page. Without protection, a script with an embedded virus can take control of the browser, making the user think they are interacting directly with the web when in fact their web passwords are being forwarded to an attacker. This design pattern — extensible applications running third party scripts — occurs in many different domains. Applications become more powerful and more widely used if third party developers and users can customize them, but that raises the issue of how to protect the application itself from rogue extensions. In this chapter we focus on how the operating system kernel against rogue applications, but the principles also apply at the application level.

A *process* is the abstraction for protection provided by the operating system kernel: the execution of an application program with restricted rights. A process needs permission from the operating system kernel before accessing the memory of any other process, before reading or writing to the disk, before changing hardware settings, and so forth. In other words, a process's access to hardware is mediated and checked by the operating system kernel. In this chapter, we explain the process concept and how the kernel implements process isolation.

Definition: **process**

A key consideration is that we need to provide protection while still running application code at high speed. The operating system kernel runs directly on the processor with unlimited rights. The kernel can perform any operation available on the hardware. What about applications? They need to run on the processor with all potentially dangerous operations disabled. To make this work, we will need a bit of assistance from hardware, which we will describe shortly. Throughout the book we will see examples of this — small amounts of carefully designed hardware can help make it much easier for the operating system to provide what users want.

Of course, both the operating system kernel and application processes running with restricted rights are in fact sharing the same machine — the same processor, the same memory, and the same disk. When reading this chapter, it is helpful to keep these two perspectives in mind: sometimes, when we are running the operating system, the system can do anything, and at other times, when we're running an application process on behalf of a user, the behavior is restricted.

Thus, a processor running an operating system is somewhat akin to someone with a split personality in charge of their own insane asylum. When running the operating system kernel, the processor is like a warden with complete access to everything. At other times, the processor runs application code in a process — the processor becomes the inmate, wearing a straightjacket locked in a padded cell by the warden, protected from harming anyone else. Of course, it is the same processor in both cases, sometimes completely trustworthy and at other

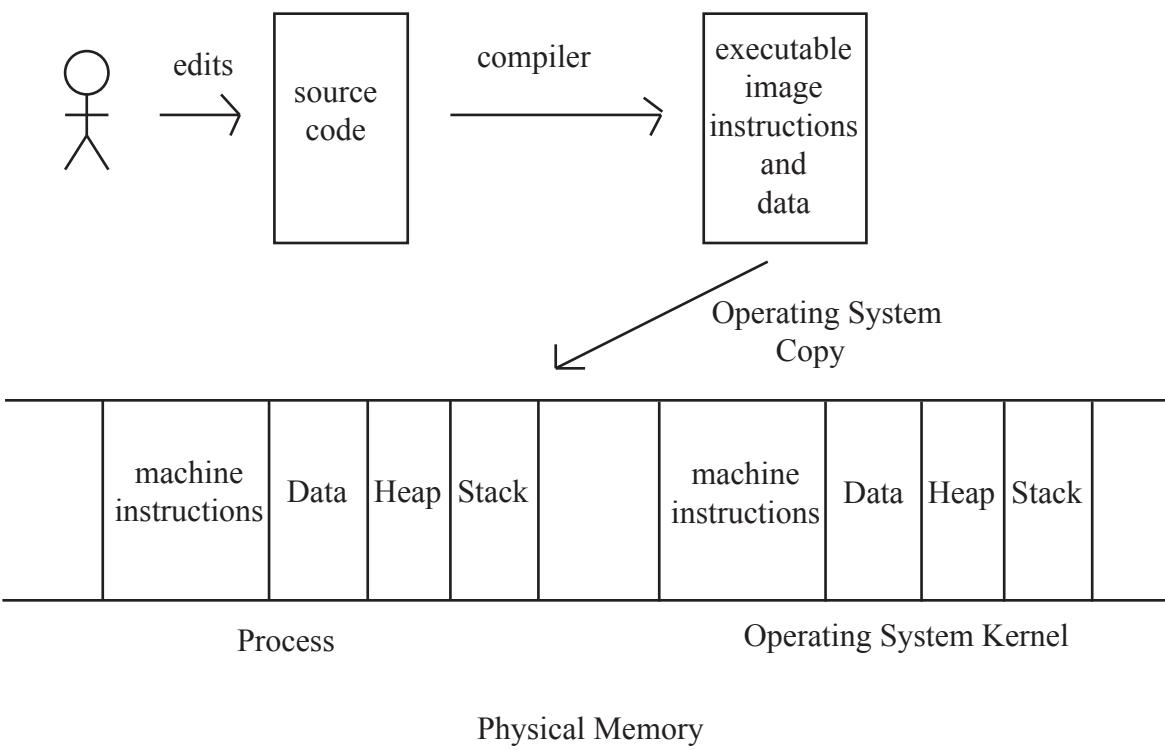


Figure 2.2: A user edits, compiles, and runs a user program. Other programs can also be stored in physical memory, including the operating system itself.

times completely untrusted.

Protection raises several important questions which we will answer in the rest of the chapter:

- **The process abstraction.** What is a process and how does it differ from a program?
- **Dual-mode operation.** How does the operating system implement processes? What hardware is needed for efficient restricted execution?
- **Safe control transfer: exceptions, interrupts, and system calls.** How do we switch safely across the boundary between processes at user-level and the kernel?
- **Booting the kernel.** What steps are needed to start running an operating system kernel, up to the point where it can create a process?

2.1 The process concept

In the model you are likely familiar with, illustrated in Figure 2.2, a programmer types up some code in some appropriately high-level language. A compiler converts that code into a sequence of machine instructions, and stores those instructions into a file, called the *executable image* of the program. The compiler also defines any static data the program needs, along with their initial values, and includes them in the executable image.

Definition: **executable image**

To start the program running, the operating system copies the program instructions and data from the executable image into physical memory. The operating system sets aside some memory for the execution stack to store the state of local variables during procedure calls. The operating system also sets aside a region of memory, called the heap, for any dynamically allocated data structures or objects the program might need. Of course, in order to copy the program into memory, the operating system itself must already be loaded into memory, with its own stack and heap.

Ignoring protection, once a program is loaded into memory, the operating system can start running the program by setting the stack pointer and jumping to the first instruction of the program. The compiler itself is just a regular program: the operating system starts the compiler by copying the compiler's executable image into memory and jumping to its first instruction.

If the user wants to run multiple copies of the same program, the operating system can do that by making multiple copies of the program's instructions, static data, heap and stack in memory. As we will describe in a later chapter, most operating systems are designed to reuse memory wherever possible: they store only a single copy of a program's instructions when multiple copies of the program are executed at the same time. Even so, a separate copy of the program's data, stack and heap are needed. For now, we will keep things simple and make a separate copy of the entire program each time we run a process.

Thus, the difference between a process and a program is that a process is an *instance* of a program, in much the same way that an object is an instance of a class in object-oriented programming. At any point in time, each program can have zero, one or more processes executing it. For each instance of a program, there is a process with its own copy of the program in memory.

The operating system keeps track of the various processes on the computer using a data structure called the *process control block*. The process control block stores all the information the operating system needs about a particular process: where it is stored in memory, where its executable image is on disk, which user asked it to start executing, what privileges the process has, and so forth.

Definition: **process control block**

Earlier, we defined a process to be the execution of a program with restricted rights. Each of these roles — execution and protection — is important enough that we will devote several chapters to each concept. In this chapter, we will

Processes, lightweight processes, and threads

The word “process”, like many terms in computer science, has evolved over time. Although you will often be able to ignore this history, it can sometimes trip up the unwary as systems built at different times will use the same word in significantly different ways.

A “process” was originally coined to mean what we now call a “thread” – some logical sequence of instructions, executing either operating system or application code. The concept of a process was developed as a way of simplifying the correct construction of operating systems, for early systems that provided no protection between application programs.

Organizing the operating system as a cooperating set of processes proved immensely successful, and soon virtually every new operating system was built this way, including systems that also provided protection against malicious or buggy user programs. At the time, almost all user programs were simple single-threaded programs with only one program counter and one stack, so there was no confusion. A process was what was needed to run a program, that is, a single sequential execution stream with a protection boundary.

As parallel computers became more popular, though, we once again needed a word for a logical sequence of instructions. A multiprocessor program can have multiple instruction sequences running in parallel, each with their own program counter, but all cooperating together within a single protection boundary. For a time, these were called “lightweight processes” (each a sequence of instructions cooperating inside a protection boundary), but eventually the word “thread” became more widely used.

This leads to the current naming convention used in almost all modern operating systems: a process executes a program, consisting of one or more threads running inside a protection boundary.

focus on protection, and so we will limit our discussion to simple processes, each with one program counter, code, data, heap, and stack.

Some programs consist of multiple concurrent activities, or threads. A web browser, for example, might need to be able to receive user input at the same time it is also drawing the screen or receiving input from the network. Each of these separate activities has its own program counter and stack but operates on the same code and data as the other threads. The operating system multiplexes threads running in a process, in much the same way that the operating system multiplexes processes into physical memory. We will generalize on the process abstraction to allow multiple activities in the same protection domain in the next chapter.

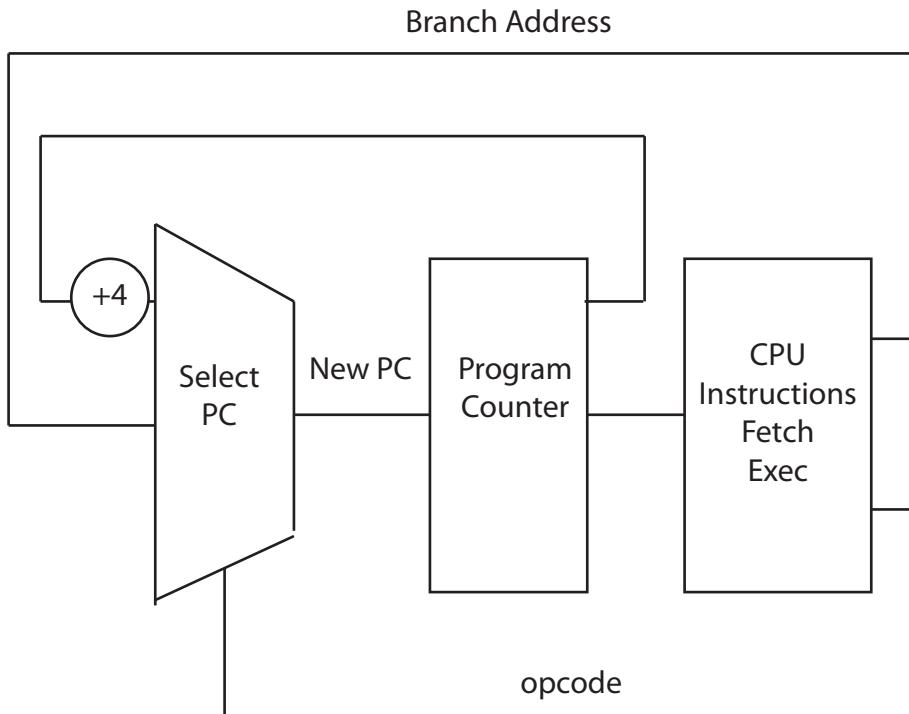


Figure 2.3: The basic operation of a CPU.

2.2 Dual-mode operation

Once a program is loaded into memory, and the operating system starts the process, the processor will fetch each instruction in turn, decode, and execute it. Some instructions compute values, say, by multiplying two registers and putting the result into another register. Some instructions read or write locations in memory. Still other instructions like branches or procedure calls change the program counter and thus determine next instruction to execute. The basic operation of a processor is illustrated in Figure 2.3.

How does the operating system kernel prevent a process from doing any harm to other processes or to the operating system itself? After all, if multiple programs are loaded into memory at the same time, what is to keep one process from overwriting another process' data structures, or for that matter, overwriting the operating system image stored on disk?

If we step back from any consideration of performance, a very simple, safe, and entirely hypothetical approach would be to have the operating system kernel simulate, step by step, every instruction in every user process. Instead of the processor directly executing instructions, each instruction in a user program would be fetched, decoded, and executed by a software interpreter. Before each

instruction is executed, the interpreter could check to see if the process had permission to do the operation in question: is it referencing part of its own memory, or someone else's? Is it trying to branch into someone else's code? Is it directly accessing the disk, or is it using the correct routines in the operating system to do so? The interpreter could allow all legal operations while halting any application that overstepped its bounds.

Now suppose we want to speed up our hypothetical simulator. Most instructions are perfectly safe, such as adding two registers together and storing the result into a third register. Can we modify the processor in some way to allow safe instructions to execute directly on the hardware?

To accomplish this, we can implement the same checks as in our hypothetical interpreter, but we do so in hardware rather than software. This is called *dual-mode operation*, represented by a single bit in the processor status register

Definition: **dual-mode operation** to signify which mode the processor is currently executing in. In *user-mode*,

Definition: **user-mode**, the processor checks each instruction before executing it to verify that the instruction is permitted to be performed by that process. (We will describe

Definition: **kernel-mode** the specific checks next.) In *kernel-mode*, the operating system executes with protection checks turned off.

Figure 2.4 shows the operation of a processor with a mode bit; the program counter and the mode control the operation of the processor. In turn, the mode bit is modified by some instructions, in the same way that the program counter is modified by some instructions.

What hardware is needed to allow the operating system kernel to protect applications and users from one another, yet also allow user code to run directly on the processor? At a minimum, the hardware must support three things:

- **Privileged instructions.** All potentially unsafe instructions are prohibited when executing in user-mode.
- **Memory protection.** All memory accesses outside of a process's valid memory region are prohibited when executing in user-mode.
- **Timer interrupts.** Regardless of what the process does, the kernel must have a way to periodically regain control from the current process.

In addition, the hardware must also provide a way to safely transfer control from user-mode to kernel-mode and back. As the mechanisms to do this are relatively involved, we defer the discussion of that topic to the following section of this chapter.

2.2.1 Privileged instructions

Process isolation is only possible if there is a way to limit programs running in user-mode from directly changing their privilege level. We will see later

The operating system kernel vs. the rest of the operating system

The operating system kernel is a crucial piece of an operating system, but it is only a portion of the overall operating system. In most modern operating systems, a portion of the operating system runs in user-mode as a library linked into each application. An example is library code to manage an application's menu buttons. To encourage a common user interface across applications, most operating systems provide a library of user interface widgets. Applications are free to write their own user interface of course, but most developers will choose to reuse the routines provided by the operating system. This code could run in the kernel but does not need to do so. If the application crashes, it won't matter if that application's menu buttons stop working. The library code (but not the operating system kernel) *shares fate* with the rest of the application: a problem with one has the same effect as a problem with the other.

Likewise, parts of the operating system can run in their own user-level processes. A window manager is one example. The window manager directs mouse actions and keyboard input that occurs inside a window to the correct application, and the manager also ensures that each application modifies only that application's portion of the screen, and not the operating system's menu bar or any other application's window. Without this, a malicious application could grab user input to itself, potentially inducing the user to disclose their password to the application, allowing it to take control of the machine.

Why not include the entire operating system — the library code and any user-level processes — in the kernel itself? While that might seem more natural, one reason is that it is often easier to debug user-level code than kernel code. The kernel can use low-level hardware to implement debugging support such as breakpoints and single step for user-level code; to single step the kernel requires an even lower-level debugger running underneath the kernel. The difficulty of debugging operating system kernels was the original motivation behind the development of virtual machines.

Further, the kernel must be trusted, as it has the full power of the hardware. Any error in the kernel may corrupt the disk, the memory of some unrelated application, or simply crash the system. By separating out code that does not need to be in the kernel, the operating system can become more reliable — a bug in the window system is bad enough, but it would be even worse if it could corrupt the disk. This is an illustration of the *principle of least privilege*, that security and reliability are enhanced if each part of the system has exactly the privileges it needs to do its job, and no more.

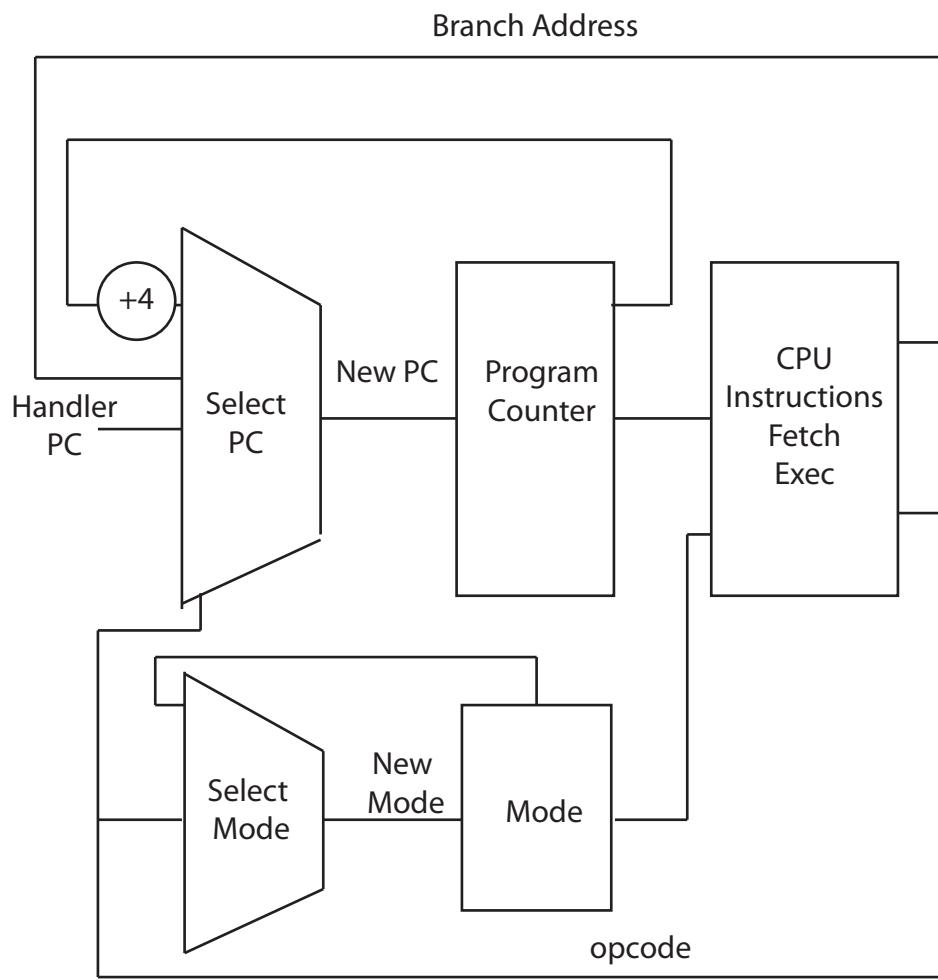


Figure 2.4: The operation of a CPU with kernel and user modes.

that processes can indirectly change their privilege level by executing a special instruction called a *system call* to transfer control into the kernel at a fixed location specified by the operating system. Other than trapping into the operating system kernel (that is, in effect, becoming the kernel) at these fixed locations, an application process cannot be allowed to change its privilege level.

Other instructions are also prohibited to application code. The application cannot be allowed to change the set of memory locations it can access; as we will see shortly, limiting an application to being able to access only its own memory is essential to preventing an application from either intentionally, or accidentally, corrupting or misusing the data or code from other applications or the operating system itself. Another limitation on applications is that they cannot disable processor interrupts, for reasons that we will also describe shortly.

Instructions available in kernel-mode, but not in user-mode, are called, naturally enough, *privileged instructions*. The operating system kernel needs to be able to execute these instructions to be able to do its work — it needs to be able to change privilege levels, adjust memory access, and disable and enable interrupts. But if these instructions were available to applications, then a rogue application would in effect have the power of the operating system kernel.

Definition: **privileged instructions**

Thus, while application programs can use only a subset of the full instruction set, the operating system executes in kernel-mode with the full power of the hardware.

What happens if an application attempts to access memory it shouldn't or attempts to change its privilege level? Such actions cause a processor *exception*. Unlike taking an exception in a programming language where the exception is handled by the language runtime, a processor exception causes the processor to transfer control to an exception handler in the operating system kernel. (We will describe in detail how exception handling works a bit later in the chapter, as exceptions can occur for many reasons beyond privilege violations.) Usually, the operating system kernel simply halts the process on privilege violations, as it often means that the application's code has encountered a bug.

Memory protection

In order to run an application process, both the operating system and the application must be resident in memory at the same time. The application needs to be in memory in order to execute, while the operating system needs to be in memory to be able to start the program, as well as to handle any system calls, interrupts, or exceptions. More generally, there are often several application processes with code and data stored in memory; for example, you may read email, download songs, Skype, instant message, and browse the web at the same time.

To make this memory sharing work, the operating system must be able to

configure the hardware so that each application process can read and write only its own memory and not the memory of the operating system or any other application. If an application could modify the operating system kernel’s code or data, it could potentially do so in a way that would give the application complete control over the machine. For example, it could change the login program to give the attacker full system administrator privileges. While it might seem that read-only access to memory is harmless, recall that we want to provide both security and privacy. Kernel data structures — such as the file system buffer — may contain data that is private to one user and not permitted to be seen by another. Likewise, user passwords may be stored in kernel memory while they are being verified.

How does the operating system prevent a user program from accessing parts of physical memory? We will discuss a wide variety of different approaches in a later chapter, but early computers pioneered a very simple mechanism that was sufficient for providing protection. We describe it as an illustration of the general principle.

Definition: base and bounds In these systems, a processor has two extra registers, called *base* and *bounds*. The base specifies the start of the process’s memory region, while the bound gives its length. This is illustrated in Figure 2.5. These registers can only be changed by privileged instructions, that is, by the operating system executing in kernel-mode. User-level code cannot change their values.

Every time the processor fetches an instruction, the address of the program counter is checked to see if it lies between the base and the bound registers. If so, the instruction fetch is allowed to proceed; otherwise, the hardware raises an exception, the program is halted, and control is transferred back to the operating system kernel. Although it might seem extravagant to do two extra comparisons for each instruction, the value of memory protection makes it worth the cost. In fact, we will see much more sophisticated and “extravagant” memory protection schemes later.

Likewise, for instructions that read or write data to memory, each memory reference is also checked against the base and bounds registers, and an exception generated if the boundaries are violated. Complex instructions, such as the x86 block copy instruction, must check every location touched by the instruction, to ensure that the application does not inadvertently or maliciously read or write to a buffer that starts in its own region but that extends into the kernel’s region. Doing so might enable the application to read or overwrite key parts of the operating system code or data and thereby gain control of the physical hardware.

The operating system kernel executes without the base and bounds registers, allowing it to access any memory on the system — the kernel’s memory or the memory of any application process running on the system. Because applications can only touch their own memory, the kernel must explicitly copy any input or output into or out of the application’s memory region. For example, a simple

program might print “hello world”. The kernel must copy the string out of the application’s memory region into the screen buffer.

Although the base and bounds mechanism is sufficient for implementing protection, it is unable to provide some important features:

- **Expandable heap and stack.** With a single base and bound per process, the memory allocated to a process’s heap and stack must be fixed at the time the program is loaded into physical memory.
- **Memory sharing.** Base and bounds does not allow memory to be shared between different processes, as would be useful for sharing code between multiple processes running the same program.
- **Non-relative memory addresses.** For base and bounds, the location of procedures and data within a program are determined at runtime, when the program is copied from disk to memory. Thus, any instruction with an absolute physical address (such as a procedure call) would need to be changed as part of loading a program into memory.
- **Memory fragmentation.** Once a program has started, it becomes nearly impossible to relocate it, as its physical addresses may be stored in registers or on the program’s execution stack, for example, as return addresses. Over time, as applications start and finish at irregular times, if memory cannot be re-compacted, it will become increasingly fragmented. Potentially, memory fragmentation may reach a point where there is not enough contiguous space to start a new process, even when there is enough memory in aggregate.

For these reasons, most modern processors introduce a level of indirection, called *virtual addresses*. With virtual addresses, every process’s memory starts at the same place, e.g., zero. Each process thinks that it has the entire machine to itself, although obviously that is not the case in reality. The hardware translates these virtual addresses to the physical location of the process in memory, e.g., by adding the base register to every virtual address.

Definition: **virtual addresses**

In practice, systems provide even more complex algorithms for translating between virtual and physical addresses. As we will see in later chapters, this level of indirection provides an enormous amount of flexibility for the operating system to efficiently manage its physical memory. For example, virtual addresses can allow the heap and the stack to start at separate ends of the virtual address space, so that they can grow according to the particular needs of each program. We illustrate this in Figure 2.6. Note that if the stack or heap grow beyond their initially allocated regions, the operating system can assign them a larger slot in physical memory, copy the stack or heap to the new location in physical memory, but leaving them in the same location in virtual memory. The copy is then completely transparently to the user process. We will discuss this in more depth in a later chapter.

As we will see, the operating system kernel can often have many different concurrent activities, each with its own execution stack. Because we need to grow these stacks dynamically, many kernels also use virtual addresses for at least some of their data.

Here is a simple test to verify that your computer supports virtual addresses. Suppose we write a program with a static variable and a procedure local variable allocated on the stack. The program updates the value of each variable, waits for a bit, and then prints the locations of the variables and their values.

If we start multiple copies of this program running simultaneously on a modern operating system with virtual addressing, each prints exactly the same result. That would be impossible if the programs were directly addressing physical memory locations. In other words, each instance of the program appears to have its own complete copy of memory, so that when it stores a value to a memory location, it is the only one that sees its changes to that location. Other processes change their own copy of the memory location. This way, a process cannot alter any other process's memory, because it has no way of referencing the other process's memory. Instead, only the kernel can read or write the memory of a process other than itself.

This is very much akin to a set of television shows, each occupying their own universe, even though they all appear on the same television. Events in one show do not (normally) affect the plot lines of other shows. Sitcom characters are blissfully unaware that Jack Bauer has just saved the world from nuclear Armageddon. Of course, just as television shows can from time to time share characters, processes can also communicate if the kernel allows it. We will discuss how this happens in a later chapter.

2.2.2 Timer interrupts

Process isolation also requires that the hardware provide a way for the operating system kernel to periodically regain control of the processor. When the operating system starts a user-level program, the process is free to execute any user-level (non-privileged) instructions it chooses, call any function in the process's memory region, load or store any value to its memory, and so forth. To the user program, it appears to have complete control of the hardware, within the limits of its memory region.

However, this too is only an illusion. If the application enters an infinite loop, or if the user simply becomes impatient and wants the system to stop the application, then the operating system needs to be able to regain control. Of course, the operating system needs to be able to execute instructions to be able to decide if it should stop the application, but if the application controls the processor, the operating system by definition is not running. A similar issue occurs in more normal circumstances. Suppose you are listening to music on your computer, downloading a file, and typing at the same time. To be able

to smoothly play the music, and to respond in a timely way to user input, the operating system needs to be able regain control promptly when it needs to switch to a new task.

Almost all computer systems include a device called a *hardware timer* which can be set to interrupt the processor after a specified delay, either in time or after some number of instructions have been executed. The operating system might set the timer to expire every ten milliseconds. (Human reaction time is measured in the hundreds of milliseconds.) Resetting the timer is a privileged operation, accessible only within the kernel, so that the user-level process cannot inadvertently or maliciously disable the timer.

Definition: **hardware timer**

When the timer interrupt occurs, control is transferred by the hardware from the user process to the operating system kernel running in kernel-mode. Other hardware interrupts, such as to signal the processor that an I/O device has completed its work, likewise cause control to be transferred from the user process to the operating system kernel. A timer or other interrupt does not imply that the program has done anything wrong — in most cases, after resetting the timer, the operating system will resume execution of the process, with the mode, program counter and registers set back to the values they had immediately before the interrupt occurred. We will discuss the hardware and operating system kernel mechanisms for implementing interrupts in the next section.

The processor status register and privilege levels

Conceptually, the kernel/user mode is a one bit register. When set to 1, the processor is in kernel mode and can do anything. When set to 0, the processor is in user mode and is restricted. On most processors, the kernel/user mode is stored in the *processor status register*. This register contains flags that control the operation of the processor. The register is typically not directly accessible to application code. Rather, flags are set or reset as a by product of executing instructions. For example, the status register is automatically saved to memory by hardware during an interrupt, because executing instructions during the interrupt will overwrite its contents.

The kernel/user mode bit is one flag in the processor status register, set whenever the kernel is entered and reset whenever the kernel is exited. Other flags include condition codes, set as a side effect of arithmetic operations, to allow a more compact encoding of conditional branch instructions. Still other flags can specify whether the processor is executing with 16-bit, 32-bit, or 64-bit addresses. The specific contents of the processor status register is processor architecture dependent.

Some processor architectures, including the Intel x86, support more than two privilege levels in the processor status register (the x86 supports four privilege levels). The original reason for this was to allow the operating system kernel to be separated into layers: a core with unlimited access to the machine, while other portions of the operating system would be restricted from certain operations, but with more power than completely unprivileged application code. This way, bugs in one part of the operating system kernel might not crash the entire system. However, to our knowledge, neither MacOS, Windows, nor Linux make use of this feature.

A potential future use for multiple privilege levels is to simplify running an operating system as an application, or virtual machine, on top of another operating system. Applications running on top of the virtual machine operating system would run at user-level, the virtual machine would run at some intermediate level, while the true kernel would run in kernel-mode. Of course, with only four levels, this doesn't work for a virtual machine running on a virtual machine running on a virtual machine. For the purposes of our discussion, we will assume the simpler case of two levels of hardware protection.

MS/DOS and memory protection

As an illustration of the power of memory protection, MS/DOS was an early Microsoft operating system that did not provide memory protection. Instead, user programs were permitted to read and modify any memory location in the system, including operating system data structures. While this was seen as acceptable for a personal computer that was only ever used by a single person at a time, there were a number of downsides. One obvious problem was system reliability: application bugs frequently crashed the operating system or corrupted other applications. The lack of memory protection also made the system more vulnerable to computer viruses.

Over time, some applications were written to take advantage of the ability to change operating system data structures, for example, to change certain control parameters or to directly manipulate the frame buffer for controlling the display. As a result, changing the operating system became quite difficult; either the new version couldn't run the old applications, limiting its appeal, or it needed to leave these data structures in precisely the same place as they were in the old version. In other words, memory protection is not only useful for reliability and security, it is also helpful as a way of enforcing a well-defined interface between applications and the operating system kernel, to aid future evolvability and portability.

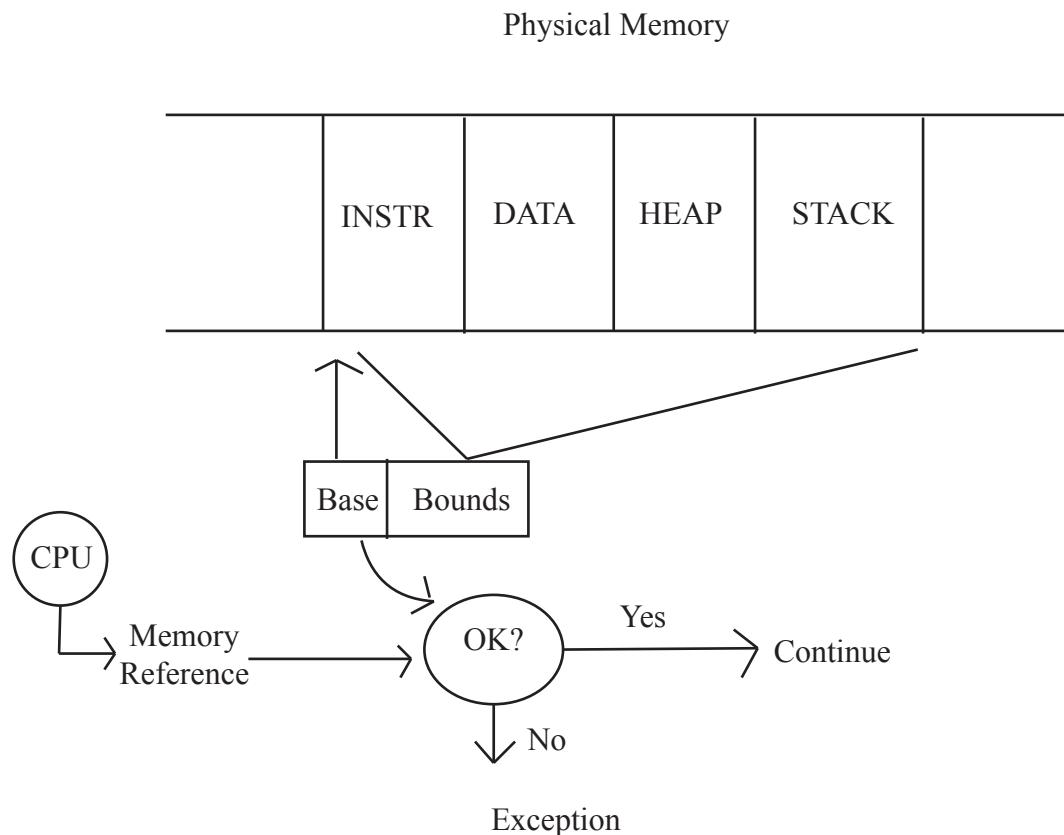


Figure 2.5: Base and bounds memory protection.

Memory-mapped devices

On most computers, the operating system controls input/output devices such as the disk, network, or keyboard by reading and writing to special memory locations. Each device monitors the memory bus for the address assigned to that device, and when it sees its address, the device triggers the desired I/O operation. The operating system sets things so that it can access these special memory locations, but user-level processes cannot. Thus, memory protection has the added advantage of limiting direct access to input/output devices by user code. By limiting each process to just its own memory locations, the kernel prevents it from directly reading or writing to the disk controller or other devices. This way, a buggy or malicious application cannot modify the operating system's image stored on disk, and a user cannot gain access to another user's files without first going through the operating system to check file permissions.

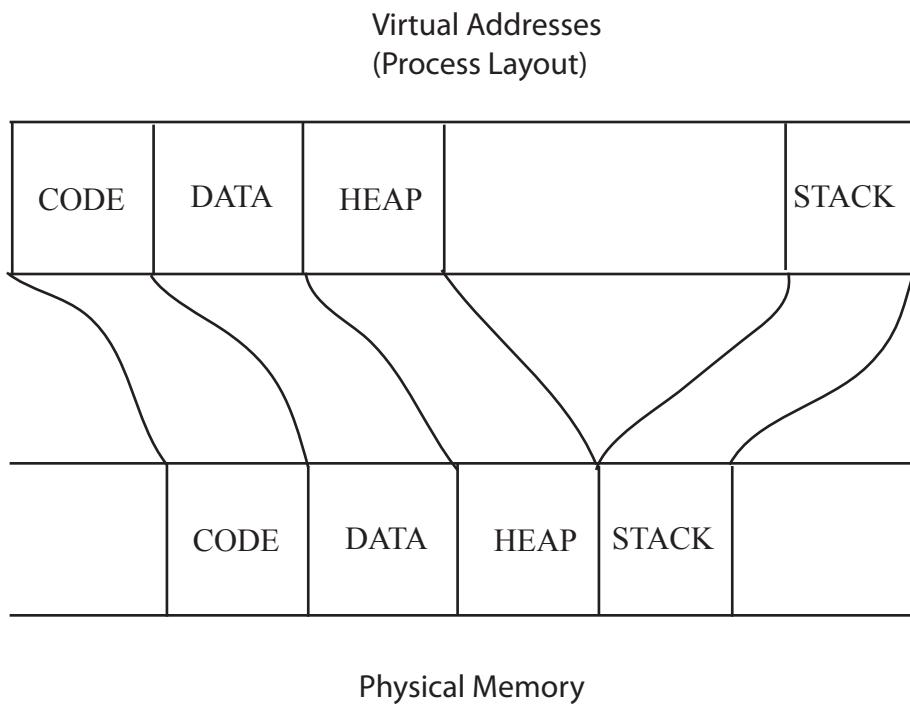


Figure 2.6: Virtual addresses allow the stack and heap regions of a process to grow independently.

```

int staticVar = 0;           /* a static variable */
main() {
    int localVar = 0;        /* a procedure local variable */

    staticVar += 1;
    localVar += 1;
    sleep(10);   /* this causes the program to wait for 10 seconds */
    printf ("static address: %x, value: %d\n", &staticVar, staticVar);
    printf ("procedure local address: %x, value: %d\n", &localVar, localVar);
}

> static address: 5328, value: 1
> procedure local address: ffffffe2, value: 1

```

MacOS and preemptive scheduling

Until 2002, Apple's MacOS operating system lacked the ability to force a process to yield the processor back to the kernel. Instead, all application programmers were told to design their systems to periodically call into the operating system to check if there was other work to be done. The operating system would then save the state of the original process, switch control to another application, and return back only when it again became the original process's turn. This has a drawback: if a process fails to yield, e.g., because it has a bug and enters an infinite loop, the operating system kernel has no recourse. The user needed to reboot the machine to give control back to the operating system. This happened frequently enough that it was given its own name: the "spinning cursor of death."

Exercises

1. We mentioned that for the “Hello world” program, the kernel must copy the string from the user program into the screen memory. Why must the screen’s buffer memory be protected? Explain what might happen if a (malicious) application could alter any pixel on the screen, not just those within its own window.
2. For each of the three mechanisms for supporting dual mode operation — privileged instructions, memory protection, and timer interrupts — explain what might go wrong without that mechanism, assuming the system still had the other two.
3. Suppose we had a perfect object-oriented language and compiler, so that only an object’s methods could access the internal data inside an object. If the operating system only ran programs written in that language, would it still need hardware memory address protection?
4. Suppose you are tasked with designing the security system for a new web browser that supports rendering web pages with embedded web page scripts. What checks would you need to ensure that executing buggy or malicious scripts could not corrupt or crash the browser?

2.3 Safe control transfer

Once the kernel has placed a user process in a carefully constructed sandbox, the next question is how do we safely transition from executing a user process to executing the kernel and the reverse. These transitions are not rare events. A high-performance web server, for example, might switch between user-mode and kernel-mode hundreds or thousands of times per second. Thus, the mechanism needs to be both fast and safe, leaving no room for a malicious or buggy program to intentionally or inadvertently corrupt the kernel.

2.3.1 User to kernel mode

We first focus on transitions from user-mode to kernel-mode; as we’ll see, transitioning in the other direction works by “undo”ing the transition from the user process into kernel.

There are three reasons for why the kernel will take control from a user process: exceptions, interrupts, and system calls.

- **Exceptions.** A processor *exception* is any unexpected condition caused by user program behavior. On an exception, the hardware will stop the currently executing process and start running the kernel at a specially designated exception handler. As we mentioned earlier, a processor exception will be triggered whenever a process attempts to perform a privileged instruction or accesses memory outside of its own memory region. Other exceptions occur when a process divides an integer by zero, accesses a word of memory with a non-aligned address, attempts to write to read-only memory, and so forth. In these cases, the operating system simply halts the process and returns an error code to the user.
Definition: exception

Exceptions can also be triggered by a number of other, more benign, program events. For example, to set a breakpoint in a program, the kernel replaces the machine instruction in memory with a special instruction that invokes an exception. When the program reaches that point in its execution, the hardware switches into the kernel. The kernel restores the old instruction and transfers control to the debugger. The debugger can then examine the program's variables, set a new breakpoint, and resume the program at the instruction causing the exception.

Definition: **interrupt**

- **Interrupts.** An *interrupt* is an asynchronous signal to the processor that some external event has occurred that may require its attention. An interrupt operates much as an exception does: the processor stops the currently executing process and starts running in the kernel at a specially designated interrupt handler. Each different type of interrupt requires its own handler. For timer interrupts, the handler checks if the current process is being responsive to user input, to detect if the process has gone into an infinite loop. The timer handler can also periodically switch the processor to run a different process, to ensure each process gets a turn. If there is no other work to do, the timer handler resumes execution at the interrupted instruction, transparently to the user process.

In addition to timer events, interrupts are also used to inform the kernel of the completion of input/output requests. For example, the mouse device hardware will trigger an interrupt every time the user moves or clicks on the mouse. The kernel in turn will notify the appropriate user process — the one the user was mousing across. Virtually every input/output device — the Ethernet, WiFi, hard disk, thumb drive, keyboard, mouse — generates an interrupt whenever some input arrives for the processor and whenever a request completes.

Definition: **polling**

An alternative to interrupts is *polling*: the kernel could loop, checking each input/output device to see if an event has occurred which required handling. Needless to say, if the kernel is polling, it is not available to run user-level code.

Interprocessor interrupts are another source of interrupts. On a multiprocessor, a processor can cause an interrupt to occur on any of the other processors. The kernel uses these interrupts to coordinate actions across

Exceptions and virtualization

Exceptions are a particularly powerful tool for virtualization — the emulation of hardware that does not actually exist. As one example, it is common for different versions of a processor architecture family to support some parts of the instruction set and not others, such as when an inexpensive low power processor does not support floating point operations. At some cost in performance, the operating system can use exceptions to make the difference completely transparent to the user process. When the program issues a floating point instruction, an exception is raised, trapping into the operating system kernel. Instead of halting the process, the operating system can *emulate* the missing instruction, and on completion, return to the user process at the instruction immediately after the one that caused the exception. This way, the same program binary can run on all the different versions of the processor.

More generally, exceptions are used to transparently emulate a virtual machine. When a guest operating system is running as a user-level process on top of an operating system, it will attempt to execute privileged instructions as if it were running on physical hardware. These instructions will cause privilege violations, trapping into the host operating system kernel. To maintain the illusion of physical hardware, the host kernel then performs the requested instruction of behalf of the user-level virtual machine, and restarts the guest operating system at the instruction immediately following the one that caused the exception.

As a final example, exceptions are a key building block for memory management. With most types of virtual addressing, the processor can be set up to take an exception whenever it reads or writes inside a particular virtual address range. This allows the kernel to treat addressing as *virtual* — a virtual address need not always correspond to a physical memory location. Whenever the program touches a missing address, the operating system takes an exception and fills in the data from disk before resuming the program. In this way, the operating system can execute programs that require more memory than can ever be physically on the machine at any one time.

the multiprocessor; for example, if one processor takes a fatal exception, the kernel will normally send an interrupt to stop any of the other processors who might be running the failed program.

- **System calls.** User processes can also transition into the operating system kernel voluntarily, to request that the kernel do some operation on the user's behalf. A *system call* is any procedure provided by the kernel that can be called from user-level. Most processors implement system calls using a special *trap* instruction. However, a trap instruction is not strictly required; we can voluntarily cause a trap by executing any instruction that causes an exception (e.g., one with an invalid opcode). As with an interrupt or an exception, the trap instruction changes the processor mode from user to kernel and starts executing in the kernel at a pre-defined handler. As we will explain shortly, to protect the kernel from

Definition: **system call**

Buffer descriptors and high-performance I/O

In early computer systems, the key to good performance was to keep the processor busy; particularly for servers, the key to good performance today is to keep I/O devices such as the network and disk device busy. Neither Internet nor disk bandwidth has kept pace with the rapid improvement in processor performance over the past four decades, leaving them relatively more important to system performance.

If only one network or disk request is outstanding at a time, interrupt handling can be the limiting factor. When a device completes a request, it raises an interrupt, triggering the operating system to make a new request. In the meantime, while the processor is handling the interrupt, the device is idle, when it could be putting the next request on the network or sending it to disk.

The solution in modern systems is for the operating system to set up a circular queue of requests, one for incoming and outgoing, for each device to handle. Each entry in the queue, called a *buffer descriptor*, specifies one I/O operation: where to find the buffer to put the incoming data or fetch the outgoing data, whether to take an interrupt on every packet or only every few packets, and so forth. This decouples the processor and the I/O devices, so that each only needs to keep up with the average rate of I/O events.

Buffer descriptors are stored in memory, accessed by the device using DMA (direct memory access). An implication is that each logical I/O operation can involve several DMA requests: one to download the buffer descriptor from memory into the device, then to copy the data in or out, and then to store the success/failure of the operation back into buffer descriptor.

misbehaving user programs, it is essential that applications trap only to a pre-defined address — they can *not* be allowed to jump to arbitrary places in the kernel.

Operating systems provide a substantial number of system calls. Examples include ones to establish a connection to a web server, to send or receive packets over the network, to create or delete files, to read or write data into files, and to create a new user process. To the user program, these are called just like normal procedures, with parameters and return values. Like any good abstraction, the caller only needs to be concerned with the interface; they do not need to know that the routine is actually being implemented by the kernel rather than by a library. The kernel handles all of the details of checking and copying arguments, performing the operation, and copying any return values back into the process's memory. When the kernel is done with the system call, it resumes user-level execution at the instruction immediately after the trap.

2.3.2 Kernel to user mode

Just as there are several different causes for transitions from user-mode to kernel-mode, there are also several causes for transitions from kernel-mode to user-mode:

- **New process.** To start a new process, the kernel copies the program into memory, sets the program counter to be the first instruction of the process, sets the stack pointer to be the base of the user stack, and switches to user-mode.
- **Resume after an exception, interrupt or system call.** When the kernel finishes handling the request, it resumes execution of the interrupted process by restoring its program counter, restoring its registers, and changing the mode back to user-level.
- **Switch to a different process.** In some cases, such as on a timer interrupt, the kernel will decide to switch to running a different process than the one that had been running before the interrupt, exception, or system call. Since the kernel will eventually want to resume the old process, the kernel needs to save the process's state — its program counter, registers, and so forth — in the process's control block. The kernel can then resume a different process, by loading its state — its program counter, registers, and so forth — from the process's control block into the processor, and then switching to user-mode.
- **User-level upcall.** Many operating systems provide user programs the ability to receive asynchronous notification of events. The mechanism, which we will describe shortly, is very similar to kernel interrupt handling, except at user-level.

2.3.3 Safe mode switch

Whether transitioning from user to kernel mode, or in the opposite direction, care must be taken to ensure that a buggy or malicious user program cannot corrupt the kernel. Although the basic idea is simple, the low-level implementation can be a bit gnarly: we need the processor to save its state and switch what it is doing, all while it continues to execute instructions that might alter the state that is in the process of saving. This is akin to rebuilding a car's transmission while it barrels down the road at 60 mph.

The context switch code must be carefully crafted, and it relies on some amount of hardware support. To avoid confusion and reduce the possibility of error, most operating systems have a common sequence of instructions for entering the kernel — whether due to interrupts, exceptions or system calls — and a common sequence of instructions for returning to user level, again regardless of the cause.

At a minimum, this common sequence must provide:

- **Limited entry.** To transfer control to the operating system kernel, the hardware must ensure that the entry point into the kernel is one set up by the kernel. User programs cannot be allowed to jump to arbitrary locations in the kernel. For example, the kernel code for handling a system call to read a file will first check whether the user program has permission to do so, and if not, the kernel should return an error. Without limited entry points into the kernel, a malicious program could simply jump immediately after the code to perform the check, allowing any user access to any file in the file system.
- **Atomic changes to processor state.** In user mode, the program counter and stack point to memory locations in the user process; memory protection prevents the user process from accessing any memory outside of its region. In kernel mode, the program counter and stack point to memory locations in the kernel; memory protection is changed to allow the kernel to access both its own data and that of the user process. Transitioning between the two must be done atomically, so that the mode, program counter, stack, and memory protection are all changed at the same time.
- **Transparent, restartable execution.** A running user-level process may be interrupted at any point, between any instruction and the next one. For example, the processor could have calculated a memory address, loaded it into a register, and be about to store a value to that address. The key to making interrupts work is that the operating system must be able to restore the state of the user program exactly as it was before the interrupt occurred. To the user process, an interrupt is invisible, except that the program temporarily slows down. A “hello world” program does not need to be written to understand interrupts! But an interrupt might still occur while it is running. Thus, on an interrupt, the processor saves its current state to memory, temporarily defers further events, and sets the processor to execute in kernel-mode, before jumping to the interrupt or exception handler. When the handler completes, the steps are reversed: the processor state is restored from its saved location, with the interrupted program none the wiser.

With that context, we first describe the hardware and software mechanism for handling an interrupt, exception, or trap, followed by how we can reuse this same basic mechanism as a building block for system calls, starting a new process, and user-level signals.

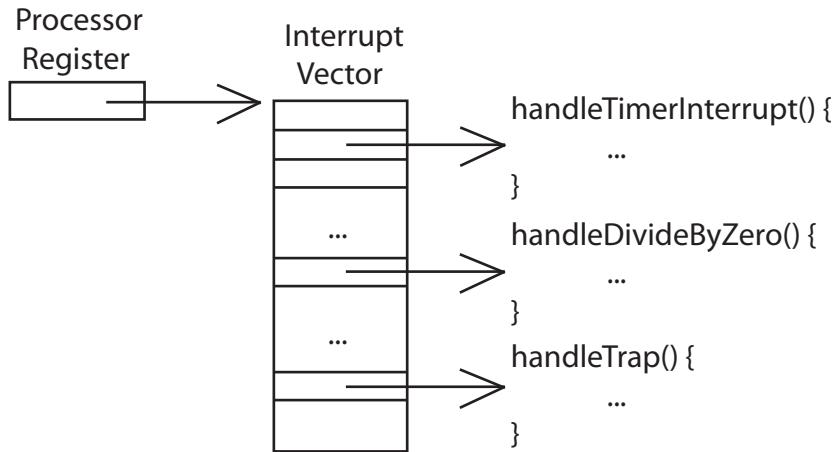


Figure 2.7: An interrupt vector identifies the code in the kernel to handle various hardware interrupts, traps, and exceptions.

Interrupt vector

When an interrupt, exception or system call trap occurs, the operating system must take different actions depending on whether the event is a divide by zero exception, a file read system call, or a timer interrupt. How does the processor know what code to run?

To identify the code to run on a context switch, the processor will include a special register that points to an area of kernel memory called the *interrupt vector*. The interrupt vector is an array of pointers, with each entry pointing to the first instruction of a different handler procedure in the kernel. As Figure 2.7 illustrates, some entries might point to various interrupt handlers such as for the timer interrupt or keyboard interrupt. Another might point to the trap handler for handling process-initiated system calls. Still other entries might point to various exception handlers like the handler for a divide-by-zero exception.

When an interrupt, trap, or exception occurs, the hardware determines which hardware device caused the interrupt, whether the trap instruction was executed, or what exception condition occurred. Thus, the hardware can select the right entry from the interrupt vector and invoke the appropriate handler.

The format of the interrupt vector is processor-specific. On the x86, for example, the interrupt vector entries 0 - 31 are for different types of hardware exceptions, entries 32 - 255 are for different types of interrupts, and by convention, entry 64 points to the system call trap handler.

Of course, the interrupt vector must be stored in protected kernel memory; otherwise a process could, for example, hijack the network by directing all network interrupts to its own code. Similarly, the hardware register that points to

the interrupt vector must be a protected register that can only be set when in kernel-mode.

Interrupt stack

Where should the interrupted process's state be saved and what stack should the kernel's code use?

On most processors, there is a special, privileged hardware register pointing to a region of kernel memory called the interrupt stack. When an interrupt, exception, or trap causes a context switch into the kernel, the hardware changes the stack pointer to point to the base of the kernel's interrupt stack. The hardware automatically saves some of the interrupted process's registers by pushing them onto the interrupt stack, before calling the kernel's handler.

When the kernel handler runs, it can then push any remaining registers onto the stack before performing the work of the handler. When returning from the interrupt, exception or trap, the reverse occurs: first we pop the registers saved by the handler, and then the hardware restores the registers it saved onto the interrupt stack, returning to the point where the process was interrupted. (In the case of returning from a system call, the value of the saved program counter needs to be incremented, so that the hardware returns to the instruction immediately after the one that caused the trap.)

You might think you could use the process's user-level stack to store its state. We use a separate kernel-level interrupt stack for two reasons.

- First, having a dedicated stack is necessary for reliability: even if the process is buggy and its stack pointer is not a valid memory address, kernel handlers can continue to work properly.
- Second, having a protected stack is necessary for security: on a multiprocessor machine if the kernel handler stored some of its variables on the user-level process's stack (for example, if the kernel handler did a procedure call), other threads in that process might read or write the kernel's return address, potentially causing the kernel to jump to arbitrary code within the kernel.

If the kernel is running on a multiprocessor system, each processor needs to have its own interrupt stack in the kernel so that, for example, we can handle simultaneous traps and exceptions across multiple processors. For each processor, the kernel allocates a separate region of memory to be used as that processor's interrupt stack.

Most operating system kernels go one step further and allocate a kernel interrupt stack for every user-level process (and as we'll see in the next chapter, every user-level thread). When a user-level process is running, the hardware

Multiprocessors and interrupt routing

On a multiprocessor, which of the various processors should take an interrupt? Some early multiprocessors dedicated a single processor (“processor 0”) to take all external interrupts. If an event required a change to what one of the other processors was doing, processor 0 could send an interprocessor interrupt to trigger that processor to switch to a new process.

For systems needing to do a large amount of input and output, such as a web server, directing all I/O through a single processor can become a bottleneck. In modern systems, interrupt routing is increasingly programmable, under control of the kernel. Often, each processor has its own, asynchronous, timer, so that different processors take interrupts at different times, reducing contention for shared data structures. Likewise, disk I/O events can be sent directly to the processor that requested the I/O operation, rather than to a random processor. Modern multicore systems can run as much as 1000 times faster if their data is already loaded into the processor cache versus if code and data needs to be transferred from some other processor.

Efficient delivery of network I/O packets is even more challenging. A high performance server might send and receive more than 100,000 packets per second, representing thousands of different connections. From a processing perspective, it is best to deliver incoming packets to the processor responsible for handling that connection; this requires the network interface hardware to demultiplex the incoming packet based on the contents of its header. Recent network controllers accomplish this by supporting multiple buffer descriptor rings for the same device, choosing which ring to use, and therefore which processor to interrupt, based on the arriving packet header.

interrupt stack points to that process’s kernel stack. Note that when a process is running at user-level, it is not running in the kernel so its kernel stack is empty.

Allocating a kernel stack per process makes it easier to switch to a new process inside an interrupt or system call handler. For example, when the timer interrupt handler runs, it might decide to give the processor to a different process. Likewise, a system call might need to wait for an I/O operation to complete, and meanwhile we would want to give the processor to someone else. With per-process stacks, to suspend a process, we simply store a pointer to its kernel stack in the process control block, and switch to running on the stack of the new process. We will describe this mechanism in much more detail in the next chapter,

Figure 2.8 summarizes the various states of a process’s user and kernel stacks:

- If the process is running on the processor in user-mode, its kernel stack is empty, ready to be used for an interrupt.
- If the process is running on the processor in kernel-mode, e.g., due to

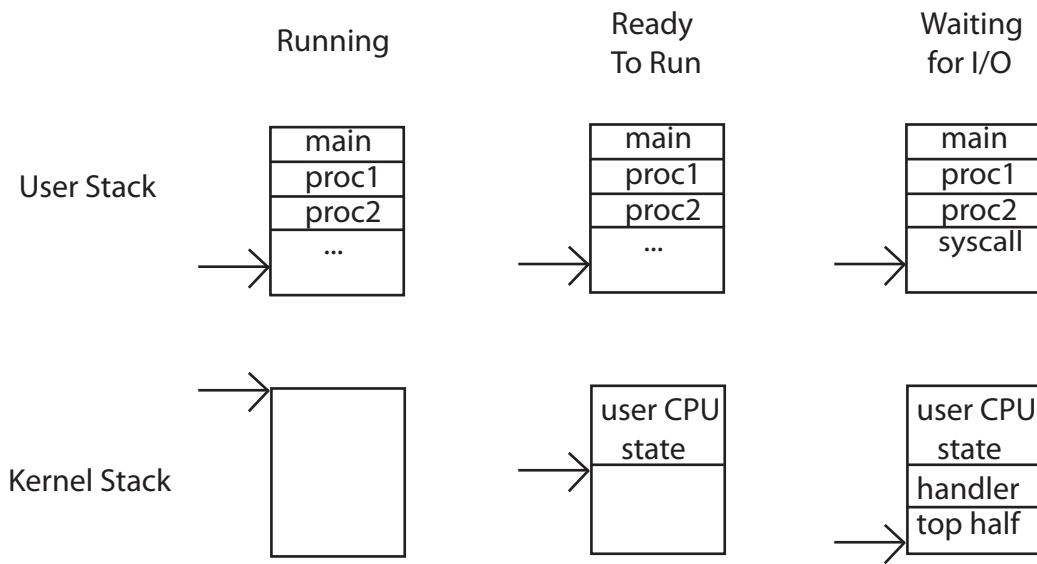


Figure 2.8: Kernel and user stacks for various states of a process.

an interrupt, exception or system call trap, its kernel stack is in use, containing the saved registers from the suspended user-level computation, as well as the current state of the kernel handler.

- If the process is available to run but is waiting for its turn on the processor, its kernel stack contains the registers and state to be restored when the process is resumed.
- If the process is waiting for an I/O event to complete, its kernel stack contains the suspended computation to be resumed when the I/O finishes.

Interrupt masking

Interrupts arrive asynchronously, so the processor might be executing either user or kernel code when an interrupt arrives. There are certain regions of the kernel (such as inside interrupt handlers themselves, or inside the CPU scheduler) where taking an interrupt might cause confusion. If we are running an interrupt handler when another interrupt occurs, we cannot just set the stack pointer to point to the base of the kernel's interrupt stack — doing so would obliterate the state of the first handler.

To simplify the design of the kernel, the hardware provides a privileged instruction to temporarily defer delivery of an interrupt until it is safe to do so. On the x86 and several other processors, this instruction is called *disable interrupts*. However, this is a misnomer: the interrupt is only deferred (masked),

UNIX and kernel stacks

In the original implementation of UNIX, kernel memory was at a premium, because main memory was roughly one million times more expensive per byte than it is today. The initial system could run with only 50KB of main memory. Instead of allocating an entire interrupt stack per process, UNIX allocated just enough memory in the process control block to store the user-level registers saved on a mode switch. This way, UNIX could suspend a user-level process with the minimal amount of memory. UNIX still needed a few kernel stacks: one to run the interrupt handler and one for every system call waiting for an I/O event to complete, but that is much less than one for every process.

Of course, now that memory is much cheaper, most systems keep things simple and allocate a kernel stack per process or thread.

and not ignored. Once a corresponding *enable interrupt* instruction is executed, any pending interrupts are delivered to the processor. The instructions to mask and unmask interrupts must be privileged — otherwise, user code could inadvertently or maliciously disable the hardware timer, allowing the machine to freeze.

If the processor is running in kernel-mode with interrupts enabled, then if an interrupt occurs, it is safe to simply use the current stack pointer rather than resetting it to the base of the exception stack. This approach can recursively push a series of handlers' states onto the stack; then, as each one completes, its state is popped from the stack and the earlier handler is resumed where it left off.

Hardware support for saving and restoring registers

The interrupted process's registers must be saved before the handler changes any of them so that the process can be restarted exactly as it left off. This is tricky because we have to save the registers without changing them in the process. Typically, the hardware provides some support to help with this process.

To make this concrete, consider the x86 architecture. First, rather than relying on the handler software to save all of the registers, when a context switch occurs the x86 hardware:

- If in user-mode, pushes the interrupted process's stack pointer onto the kernel's exception stack, and switches to the kernel stack.
- Pushes the interrupted process's instruction pointer.
- Pushes the x86 *processor status word*. The processor status word includes control bits such as whether the most recent arithmetic operation in the

Interrupt handlers: Top and bottom halves

When a machine invokes an interrupt handler because some hardware event occurred (e.g., a timer expired, a key was pressed, a network packet arrived, or a disk I/O completed), the processor hardware typically masks interrupts while the interrupt handler executes. While interrupts are masked, if another hardware event occurs, it will not trigger another invocation of the interrupt handler until the interrupt is reenabled.

Some interrupts can trigger a large amount of processing, and it is undesirable to leave interrupts masked for too long. Hardware I/O devices have a limited amount of buffering, and this can lead to dropped events if interrupts are not processed in a timely fashion. For example, the keyboard hardware will drop keystrokes if the keyboard buffer is full. Interrupt handlers are therefore often divided into a *top half* and a *bottom half*. Unfortunately, this terminology can differ a bit from system to system, so in this book we use the more common usage.

The interrupt handler's bottom half is invoked by the hardware and executes with interrupts masked. It is designed to complete quickly. The bottom half typically saves the state of the hardware device, resets it so that it can receive a new event, and notifies the scheduler that the top half needs to run. At this point, the bottom half is done, and it can re-enable interrupts and return to the interrupted task, or (if the event is high priority) switch to the top half but with interrupts enabled. When the top half runs, it can do more time consuming things, such as parsing the arriving packet, delivering it to the correct user-level process, sending an acknowledgment, and so forth.

As we will explain in a later chapter, while the bottom half must always be able to run to completion in a bounded amount of time, the top half can do operations that require waiting, and therefore can synchronize access to shared kernel data structures.

Be aware, however, that on Linux, the terms are reversed: the bottom half is called the top half and vice versa.

interrupted code resulted in a positive, negative, or zero value (needed for the correct behavior of any subsequent conditional branch instruction).

This hardware feature is needed because it is essential to save these *before* running the handler software: once the handler software starts running, the stack pointer, instruction pointer, and processor status word will be those of the handler, not those of the interrupted process.

Once the handler starts running, it can use the `pushad` instruction to save the remaining registers onto the stack. `pushad` saves the x86 integer registers; because the kernel does not typically do floating point operations, those do not need to be saved unless the kernel switches to a new process.

x86 has complementary features for restoring state: a `popad` instruction to pop an array of integer register values off the stack into the registers and an `iret` (return from interrupt) instruction that loads a stack pointer, instruction pointer, and processor status word off of the stack into the appropriate processor

registers.

Putting it all together: Mode switch on the x86

As noted above, the high level steps needed to handle an interrupt, trap, or exception are simple, but the details require some care.

To give a concrete example of how such “carefully crafted” code works, we now describe one way to implement interrupt-triggered context switch on the x86 architecture. Different operating systems on the x86 follow this basic approach, though details differ. Similarly, different architectures handle the same types of issues, but they may do so with different hardware support.

First, a bit of background on the x86 architecture. The x86 is segmented, and so pointers come in two parts: a segment, such as code, data or stack, and an offset within that segment. The current user-level instruction is based on a combination of the code segment (`cs` register plus the instruction pointer `eip`). Likewise, the current stack position is based on the stack segment `ss` and the stack pointer within the stack segment `esp`. The current privilege level is stored as the low order bits in the `cs` register, rather than the processor status word `eflags`. The `eflags` register has condition codes that are modified as a by-product of executing instructions; the `eflags` register also has other flags that control the processor’s behavior such as whether interrupts are masked or not.

When a user-level process is running, the current state of the processor, stack, and kernel interrupt vector and kernel stack is illustrated in Figure 2.9. When an exception or trap occurs, the hardware carefully saves a small amount of the interrupted thread state, as illustrated in Figure 2.10:

1. **Save three key values.** The hardware internally saves the value of the stack pointer (the x86 `esp` and `ss` registers), the execution flags (the x86 `eflags` register), and the instruction pointer (the x86 `eip` and `cs` registers).
2. **Switch onto the kernel exception stack.** The hardware then switches the stack pointer to the base of the kernel exception stack, specified in a special hardware register.
3. **Push the three key values onto the new stack.** The hardware then stores the internally saved values onto the stack.
4. **Optionally save error code.** Certain types of exceptions such as page faults generate an error code to provide more information about the event; for these exceptions, the hardware pushes this code as the last item on the stack. For other types of events, the software interrupt handler typically pushes a dummy value onto the stack so that the stack format is identical in both cases.

5. **Invoke the interrupt handler.** Finally, the hardware changes the program counter to the address of the interrupt handler procedure, specified via a special register in the processor that is accessible only to the kernel. This register contains a pointer to an array of exception handler addresses in memory. The type of interrupt is mapped to an index in this array, and the program counter is set to the value at this index.

Notice that changing the program counter to start of the interrupt handler means that the next thing that happens is that the interrupt handler software begins to run.

The first thing the handler needs to do is save the rest of the interrupted process's state—it needs to save the other registers before it changes them! So, the handler pushes the rest of the registers, including the current stack pointer, onto the stack. The x86 `pushad` instruction, which pushes the contents of all general purpose integer registers onto the stack, is convenient here.

As Figure 2.11 indicates, at this point the kernel's exception stack holds (1) the stack pointer, execution flags, and program counter saved by the hardware, (2) an error code or dummy value, and (3) a copy of all of the general registers (including the stack pointer but not the instruction pointer or `eflags` register) as they were after the hardware pushed the stack pointer, execution flags, and program counter onto the exception stack.

Once the handler has saved the interrupted thread's state to the stack, it is free to use the registers as it pleases, and it can also push additional items onto the stack. So, the handler can now do whatever work it needs to do.

When the handler completes, it can resume the interrupted process. To do this, the handler pops all of the registers saved by the software from the stack, thereby restoring all registers except the execution flags, program counter, and stack pointer (which was changed by the hardware when the hardware pushed the execution flags and program counter onto the stack.) For the x86 instruction set, the `popad` instruction is commonly used. The handler also pops the error value off the stack

Finally, the handler executes the x86 `iret` instruction to pop the program counter, execution flags, and stack pointer from the kernel's exception stack stack into their respective registers.

Thus, the interrupted thread's state has been restored to exactly what it was before the interrupt, and it continues execution, as if nothing happened.

A small but important detail occurs when the operating system returns from an exception, where the instruction is emulated by the kernel, e.g., for emulating floating point hardware. If we return back to the instruction that caused the exception, another exception will instantly recur! To prevent an infinite loop, the exception handler modifies the program counter stored at the base on the stack to point to the instruction immediately after the one causing the mode

switch. The `iret` instruction will then return to the user process at the correct location.

In the case of a system call trap, the hardware does the increment for us — the program counter for the instruction after the trap is saved on the kernel’s interrupt stack.

2.3.4 System calls

The operating system kernel constructs a restricted environment for process execution to limit the consequences of erroneous and malicious programs on the reliability of the rest of the computer system. Any time a process needs to do anything that operates outside of its protection domain, such as create a new process, read from the keyboard, or write a disk block, the process must ask the operating system to do the action on its behalf, via a system call.

System calls provide the illusion that the operating system kernel is simply a set of library routines available for use by user programs. To the user program, there are a set of system call procedures, each with arguments and return values, that can be called like any other routine. The user program need not concern itself with how the operating system implements the system call.

This requires us to define a *calling convention* for naming system calls, passing arguments, and receiving return values across the user/kernel boundary. Any convenient calling convention will do, such as putting arguments on the user stack, passing them in registers, and so forth. Typically, the operating system defines a common convention that is used by both compilers and the operating system kernel.

Once the arguments are in the right format, the user level program can issue a system call by executing the trap instruction to transfer control to the operating system kernel, as described above. System calls share much of the same mechanism for switching between user and kernel mode as interrupts and exceptions. In fact, one frequently used Intel x86 instruction to trap into the kernel on a system call is called `int`, for “software interrupt.”

Inside the operating system kernel there is a procedure to implement each system call. This procedure is exactly as if the call was made from within the kernel with one notable exception. The kernel must implement its system calls in a way that protects itself from all errors and attacks that might be launched by the misuse of the interface. One can think of this as an extreme version of defensive programming: the kernel should assume the parameters to each system call are intentionally designed to corrupt the kernel.

We meld these two views — the user program calling the system call, and the kernel implementing the system call — with a pair of stubs. A *pair of stubs* are two short procedures that mediate between two environments, in this case

Definition: **pair of stubs**

between the user program and the kernel. As we will see later, stubs are also used to mediate procedure calls between computers in a distributed system.

The system call stubs are illustrated in Figure 2.12. The user program calls the user stub in the normal way, oblivious to the fact the implementation of the procedure is in fact in the kernel. The user stub calls the trap instruction. The hardware transfers control to the kernel, vectoring to the system call handler. The handler acts as a stub on the kernel side, calling the implementation of the routine in the kernel. The return path follows the reverse sequence.

We illustrate the behavior of the user-level stub in Figure 2.13 for the x86. For each system call, the operating system provides a library routine for that system call. This library routine takes its arguments, reformats them according to the calling convention defined by the operating system kernel, and executes a trap instruction. When the kernel returns, the stub can return the result provided by the kernel. Of course, the user program need not use the library routine — it is free to trap directly to the kernel; in turn, the kernel needs to protect itself from misbehaving programs that do not format arguments correctly.

The system call calling convention is arbitrary, so here we pass arguments on the user stack, with a code indicating the type of system call in the register `%eax`. The return value comes back in `%eax` so there is no work to do on the return.

The `int` instruction operates as described earlier, saving the program counter, stack pointer, and eflags on the kernel stack, before calling the system call handler through the interrupt vector. Depending on the calling convention, the kernel handler saves any additional registers that must be preserved across function calls. It then examines the system call integer code in `%eax`, verifies that it is a legal opcode, and calls the correct stub for that system call.

The kernel stub has four tasks:

- **Locate system call arguments.** Unlike a regular procedure, the arguments to a system call are stored in user memory. In the example code above, they were stored on the process's user stack. Of course, the user stack pointer could be corrupted! Even if it is valid, it is a virtual address, not a physical address. Any address provided by a user program to a system call must be checked (to verify it is a legal address within the user domain) and converted to a physical address that the kernel can use to access the memory. In the example above, the pointer to the string representing the file name is stored on the stack, so both the stack address must be (checked and) translated, and then the value of the string pointer stored on the stack must be (checked and) translated.
- **Validate parameters.** The operating system kernel must also protect itself against malicious or accidental errors in the format or content of its arguments. A file name will normally be a zero-terminated string, but the kernel cannot trust the user code to always work correctly. The file

name may be corrupted, it may point to memory outside the application’s region, it may start inside the application’s memory region but extend beyond it, the application may not have permission to access the file, and so forth. If an error is detected, the kernel returns to the user program with an error; otherwise the kernel performs the operation on behalf of the application.

- **Copy before check.** In most cases, the kernel copies system call parameters into kernel memory before performing the necessary checks. The difficulty arises if the application can modify the user memory holding a system call parameter (such as a file name), *after* the check is performed, but *before* the parameter is used in the actual implementation of the routine. This is called a “Time of Check vs. Time of Use”, or *TOCTOU attack*. This is not a new attack — the first occurrence dates from the mid-1960’s. While it might seem that a process necessarily stops whenever it does a system call, this can be misleading. For example, if one process shares a memory region with another process, then the two processes working together can launch the TOCTOU attack. Similarly, on a multiprocessor, one processor can launch the TOCTOU attack while the other processor traps into the kernel with the system call.
- **Copy back any results.** If the system call reads data into a buffer in user memory, the stub needs to copy the data from the kernel buffer into user memory so that the program can access it. Again, the kernel must first check the user address and convert it to a kernel address before use.

Definition: **TOCTOU attack**

Putting this together, the kernel stub for the system call “open” is given in Figure 2.14.

When the system call completes, it returns back to the stub which returns back to the system call handler. At this point, the stub takes steps to pass the results of the system call back to the user process. In the example, the return value fits in a register so the stub can return directly; in the case of a file read, the stub would need to ensure that the data is placed in the buffer pointed to by the user, that is, in the user program’s memory.

In turn, the system call handler pops any saved registers (except %eax) and uses the `iret` instruction to return back to the user stub immediately after the trap, allowing the user stub to return to the user program.

2.3.5 Starting a new process

So far, we have described how we transfer control from a user-level process to the kernel on an interrupt, exception, or system call, and how the kernel resumes execution at user-level when done.

With that context, we can complete the description of how we start running at user-level in the first place. The mechanism is straightforward, if a bit

backhanded. The kernel allocates and initializes the process control block, allocates memory for the process, copies the program from disk into the newly allocated memory, and allocates both a user-level stack for normal execution and a kernel-level stack for handling system calls, interrupts and exceptions.

To start the program running, we need two additional steps:

- **Copy arguments into user memory.** When starting a program, the user sometimes gives it arguments, much like calling a procedure. For example, when you click on a file icon in Mac OS or Windows, the window manager asks the kernel to start the application associated with the file, passing it the file name to open. The system call to create a process then copies the file name from the memory of the window manager process to a special region of memory in the new process. By convention, arguments to a process are typically copied to the base of the user-level stack, and the user's stack pointer is incremented so those addresses are not overwritten when the program starts running.
- **Transfer control to user-mode.** When we start a new process, it never trapped into the kernel to save its current state. While we could special-case the code for starting a new process, most operating systems re-use the same code to exit the kernel for starting a new process and for returning from a system call. When we create the new process, we allocate it a kernel stack, and we reserve room at the bottom of the kernel stack for the initial values of its registers, program counter, stack pointer, and processor status word. To start the new program, we can then switch to the new stack and jump to the end of the interrupt handler. When the handler executes `popad` and `iret`, the processor "returns" to the start of the user program.

Finally, although you can think of a user program as starting with a call to `main`, in fact the compiler inserts one level of indirection. It generates a stub, at the location within the process's memory where the kernel will jump on process start. The stub's job is to call `main` and then if `main` returns, calls `exit`. Without this, a user program that returned from `main` would try to pop the return program counter from `main`, and since there wasn't any such address on the stack, the processor would start executing random code. `Exit` is a system call that terminates the process.

2.3.6 Upcalls

We can use system calls for most of the communication between applications and the operating system kernel. When a program needs to request some protected operation, it can trap to ask the kernel to perform the operation on its behalf. Likewise, if there is data inside the kernel that the application needs, the program can simply perform a system call to retrieve it.

If we want to allow applications to be able to implement operating system-like functionality, we need something more. For many of the reasons that operating system kernels need interrupt-based event delivery, applications can also benefit from being told when some event that deserves its immediate attention has occurred. Throughout this book, we will see this pattern repeatedly: the need to *virtualize* some part of the operating system kernel, so that applications can behave more like operating systems. We call virtualized interrupts and exceptions *upcalls*. In UNIX, they are called *signals*, and in Windows they are Definition: **upcalls** called *asynchronous events*.

Here are some cases where immediate event delivery with upcalls are useful:

- **Preemptive user-level thread package.** Just as the operating system kernel multiplexes processes onto a processor, an application may want to multiplex its tasks, or threads, onto a process. A user-level thread package can assume all of its tasks run to completion or at least cooperatively yield the processor from time to time, but another approach is to use a periodic timer upcall as a trigger to switch tasks. This is helpful to share the processor more evenly among user-level tasks. It is also helpful for stopping a runaway task, e.g., if the application needs to run third party code, such as when a web browser executes an embedded web page script.
- **Asynchronous I/O notification.** A system call is designed to wait until the requested operation is complete and then return. What if the process has other work it could have done in the meantime? One approach is *asynchronous I/O*: a system call starts the request and returns immediately; later on, the application can poll the kernel for I/O completion, or a separate notification can be sent via an upcall to the application when the I/O completes. Definition: **asynchronous I/O**
- **Interprocess communication.** Most interprocess communication can be handled with system calls — one process writes data, while the other reads it sometime later. A kernel upcall is needed if a process generates an event that needs the instant attention of another process. As an example, UNIX will send an upcall to notify a process when the debugger wants to suspend or resume the process. Another use is for logout — to notify applications that they should save file data and cleanly terminate.
- **User-level exception handling.** Earlier, we described a mechanism where hardware exceptions, such as divide by zero errors, are handled by the operating system kernel. However, many applications have their own exception handling routines, e.g., to ensure that files are saved before the application shuts down, or in some cases, to terminate an offending script, while the rest of the application continues to work. For this, the operating system needs to inform the application when it receives an exception, so the application runtime, rather than the kernel, handles the event.

- **User-level resource allocation policy.** One of the tasks of an operating system is resource allocation — deciding which users or which processes get how much CPU time, how much memory, and so forth. In turn, many applications are resource adaptive — able to optimize their behavior to differing amounts of CPU time or memory. An example is a garbage collected system like the Java runtime. Within limits, a Java process can adapt to different amounts of available memory by changing the frequency of how often it runs its garbage collector. The more memory, the less time Java needs to run its collector, speeding up execution. This only works if the operating system somehow is able to inform the process when its allocation needs to change, e.g., because some other process needs more memory or less at the moment.

We should note that upcalls from kernels to user processes are not always needed. Many applications are more simply structured around an event loop that polls for events, and then processes each event in turn. In this model, the kernel can pass data to the process by sending it events, provided they do not need to be handled immediately. In fact, until recently, Windows lacked support for the immediate delivery of upcalls to user-level programs.

We next describe UNIX signals as a concrete example of kernel support for upcalls. As shown in Figure 2.15 and Figure 2.16, most of the features in UNIX signals have a direct analogue with hardware interrupts:

- **Types of signals.** In place of hardware-defined interrupts and exceptions, the kernel defines a limited number of types of signals that can be received by a process.
- **Handlers.** Each process defines its own handlers for each signal type, much as the kernel defines its own interrupt vector table. If a process does not define a handler for a specific signal, then the kernel calls a default handler instead.
- **Signal stack.** Applications have the option to run UNIX signal handlers either on the process's normal execution stack or on a special signal stack allocated by the user process in user memory. Running signal handlers on the normal stack can reduce the flexibility of the signal handler in manipulating the stack, e.g., to cause a language-level exception to be raised.
- **Signal masking.** A UNIX signal handler automatically masks further delivery of that type of signal until the handler returns. The program can mask other signals, either all together or individually, as needed.
- **Processor state.** The kernel provides the UNIX signal handler a data structure containing the saved state of the program counter, stack pointer, and general purpose registers when the program was interrupted. Normally, when the handler returns, the kernel reloads the saved state back

into the processor to resume execution of the program. However, the handler can also modify the state to be restored, e.g., if it wants to switch to another user-level task.

The mechanism for delivering UNIX signals to user processes requires only a small modification to what we have already described for transferring control across the kernel-user boundary. For example, on a timer interrupt, the hardware and the kernel interrupt handler save the state of the user-level computation. To deliver the timer interrupt to user-level, the kernel copies that saved state to a user-level buffer, resets the saved state to point to the signal handler and signal stack, and then exits the kernel handler. The `reti` instruction will then resume execution at the signal handler, rather than the original program counter. When the signal handler returns, these steps are unwound: the processor state is copied back from the signal handler into kernel memory, and the `reti` then returns back to the original computation.

Architectural support for fast mode switches

Some processor architectures are able to execute user and kernel mode switches very efficiently, while other architectures are much slower at performing these switches.

The SPARC architecture is in the first camp. SPARC defines a set of *register windows* that operate like a hardware stack. Each register window includes a full set of the registers defined by the SPARC instruction set. When the processor performs a procedure call, it shifts to a new window, so the compiler never needs to save and restore registers across procedure calls, making them quite fast. (At a deep enough level of recursion, the SPARC will run out of its register windows; it then takes an kernel exception that saves half the windows and resumes execution. Another exception is taken when the processor pops its last window, allowing the kernel to reload the saved windows.)

Mode switches can be quite fast on the SPARC. On a mode switch, the processor switches to a different register window. The kernel handler can then run, using the registers from the new window and not disturbing the values stored in the interrupted process's copy of its registers. Unfortunately, this comes at a cost: switching between different processes is quite expensive on the SPARC, as the kernel needs to save and restore the entire register set of every active window.

The Motorola 88000 was in the second camp. The 88000 was an early pipelined architecture; for improved performance, multiple instructions were in various stages of execution at the same time. For example, one instruction might be fetching the instruction while another is doing a floating point operation and yet another is finishing a store to memory. When an interrupt or exception occurred on the 88000, the pipeline operation was suspended, and the operating system kernel was required to save and restore the entire state of the pipeline to preserve transparency.

Most modern processors with deep execution pipelines, such as the x86, instead provide *precise interrupts*: in hardware, all instructions that occur before the interrupt or exception, according to the program execution, are completed by the hardware before the interrupt handler is invoked. Any instruction is annulled if it occurs in the program after the interrupt or exception, even if the instruction is in progress when the processor detects the exception.

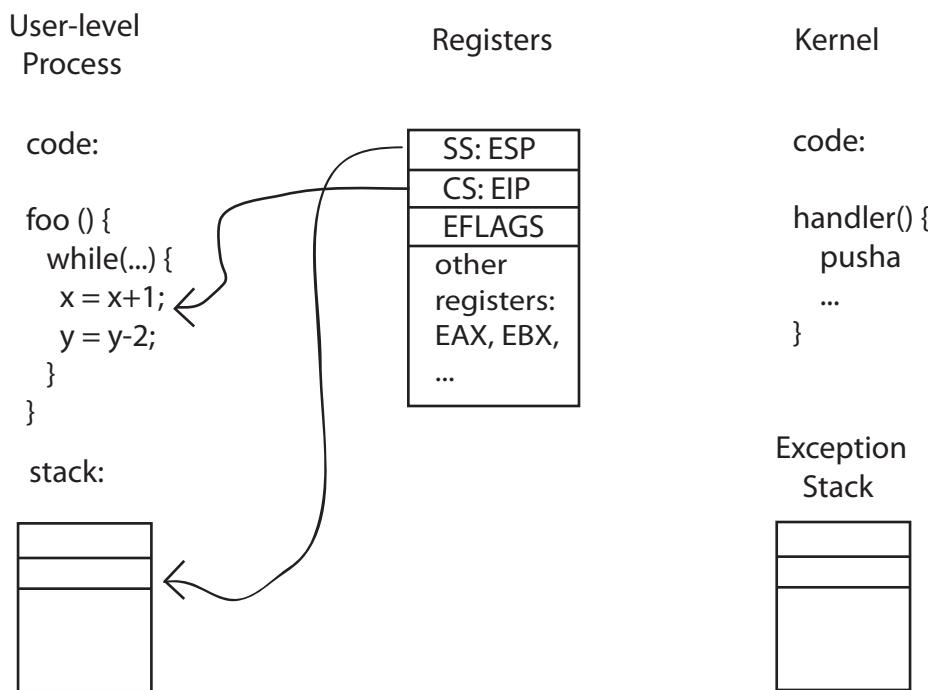


Figure 2.9: State of the system before an interrupt handler is invoked on the x86 architecture.

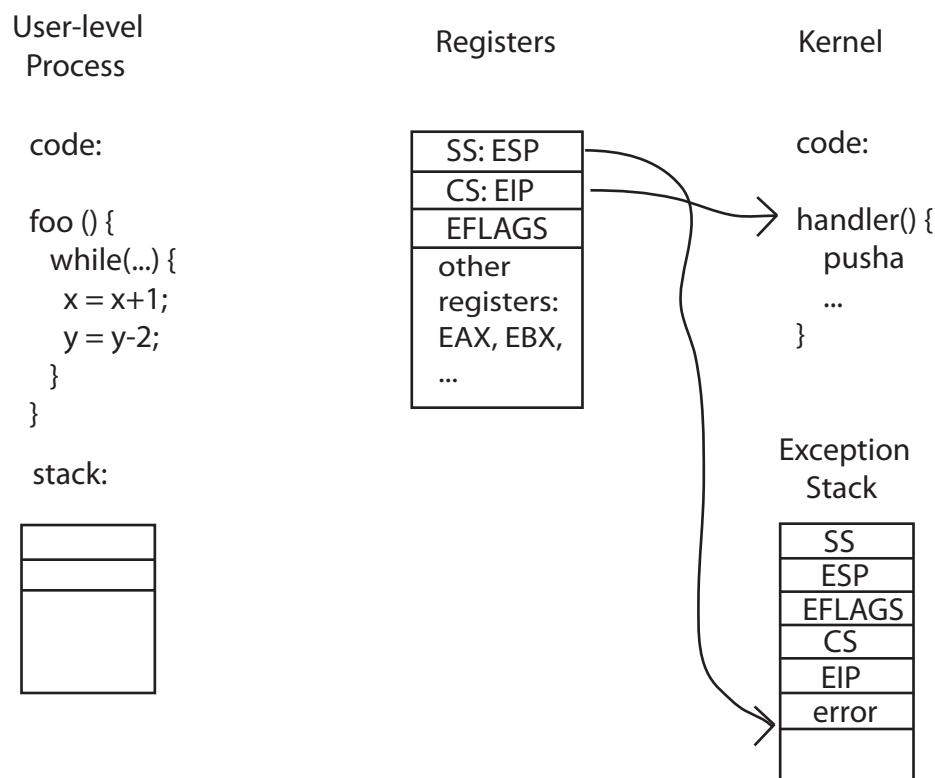


Figure 2.10: State of the system after the hardware has jumped to the interrupt handler on the x86 architecture.

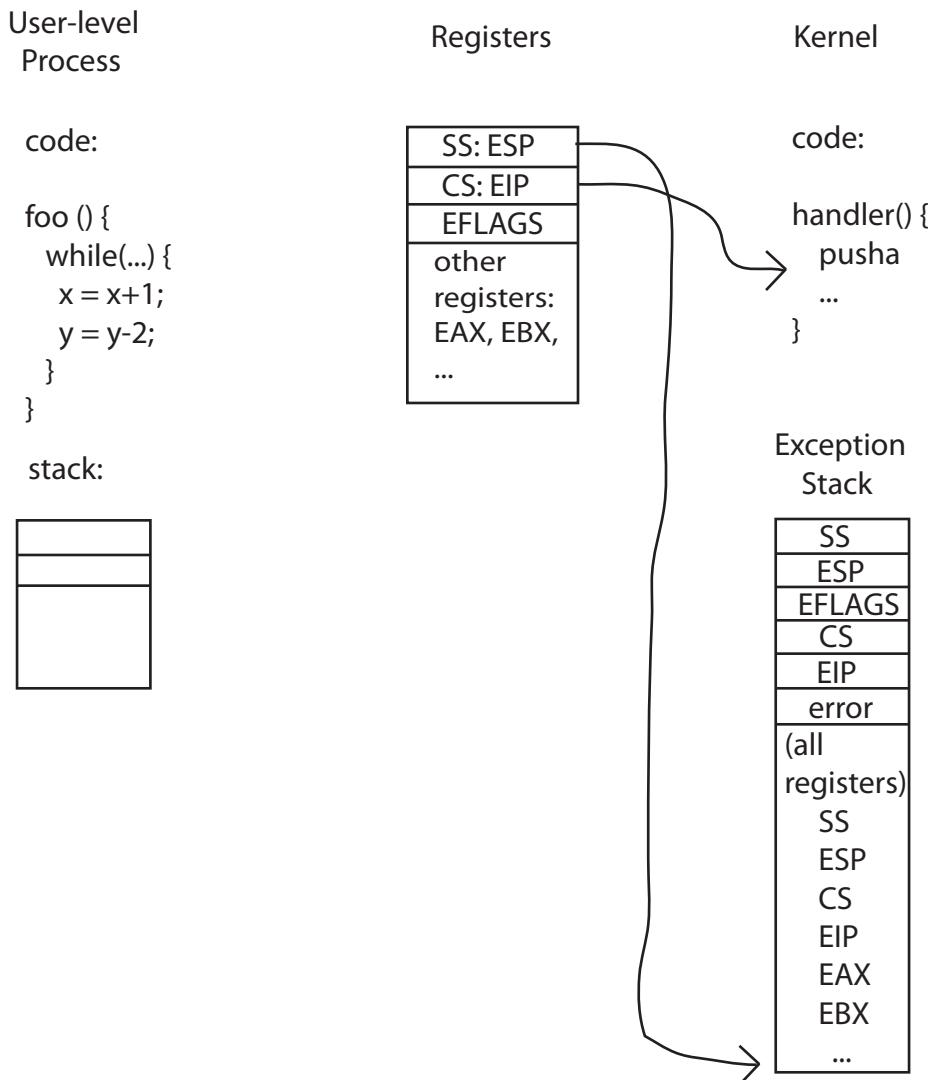


Figure 2.11: State of the system after the interrupt handler has started executing on the x86 architecture.

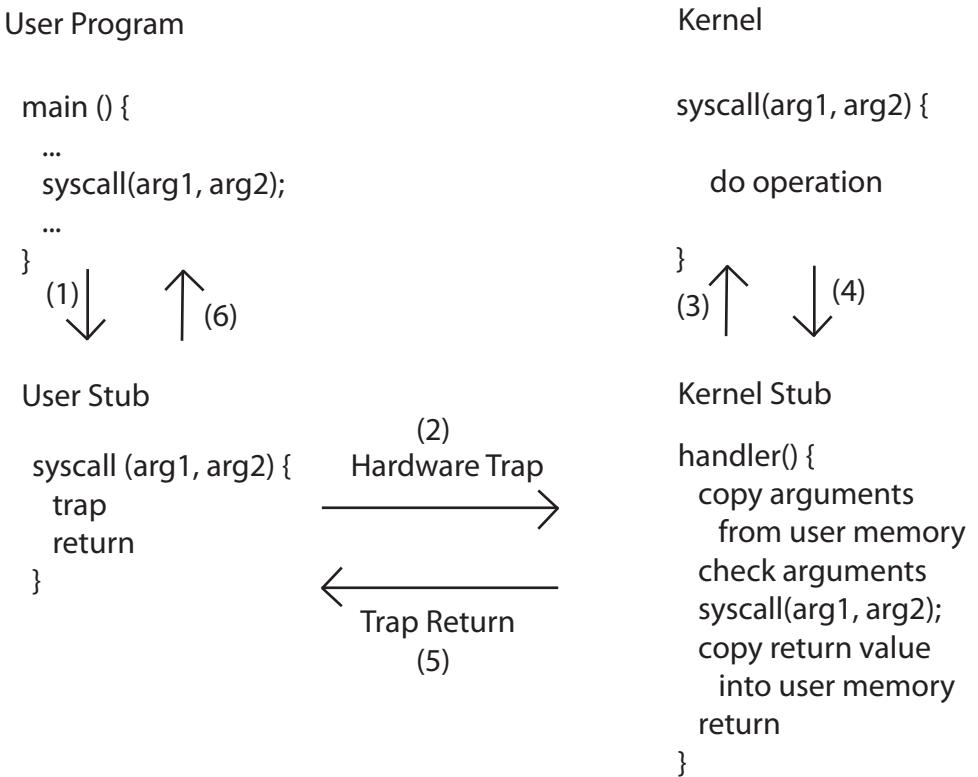


Figure 2.12: Stubs mediate between the user-level caller and the kernel implementation of system calls.

```
open:           // we assume the caller put the filename onto the stack already,
               // according to the standard calling convention for the x86
    movl #SysCall_Open, %eax      // tell the kernel which system call routine we want
    int 32                      // trap into the kernel
    ret                         // return back to caller, assumes kernel left return value in %eax
```

Figure 2.13: User level library stub to invoke the file system open system call.

```

int KernelStub_Open() {
    char *localCopy[MaxFileNameSize + 1];

    // check that stack pointer is valid and that arguments are stored at valid addresses
    if (!validUserAddressRange(userStackPointer, userStackPointer + size of arguments on stack))
        return error_code;

    // fetch pointer to file name from user stack, and convert to a kernel pointer
    filename = VirtualToKernel(userStackPointer);

    // make a local copy of the filename, inside the OS
    // this prevents the application from changing the name
    // surreptitiously, after the check, but before the read!

    // the string copy needs to check each address in the string before use
    // to make sure every address in the string is valid

    // the string copy terminates after it copies MaxFileNameSize
    // to ensure we don't overwrite our internal buffer

    if (!VirtualToStringCopy(filename, localCopy, MaxFileNameSize))
        return error_code;

    // let's make sure our local copy is null terminated
    localCopy[MaxFileNameSize] = 0;

    // we can now check if the user is permitted to access this file
    if (!UserFileAccessPermitted(localCopy, current_process))
        return error_code;

    // finally we can call the actual routine to open the file
    // this returns a file handle on success, or an error_code on failure

    return Kernel_Open(localCopy);
}

```

Figure 2.14: Stub routine for the open system call inside the kernel

```

start(arg1, arg2) {
    main(arg1, arg2);           // call user's main
    exit();                    // if main returns, call exit
}

```

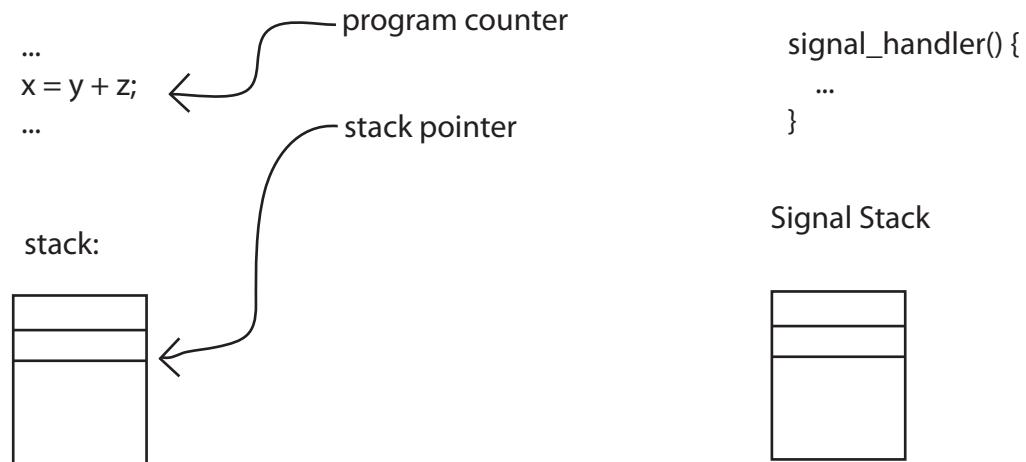


Figure 2.15: The state of the user program and signal handler before a UNIX signal. UNIX signals behave analogously to hardware exceptions, but at user-level.

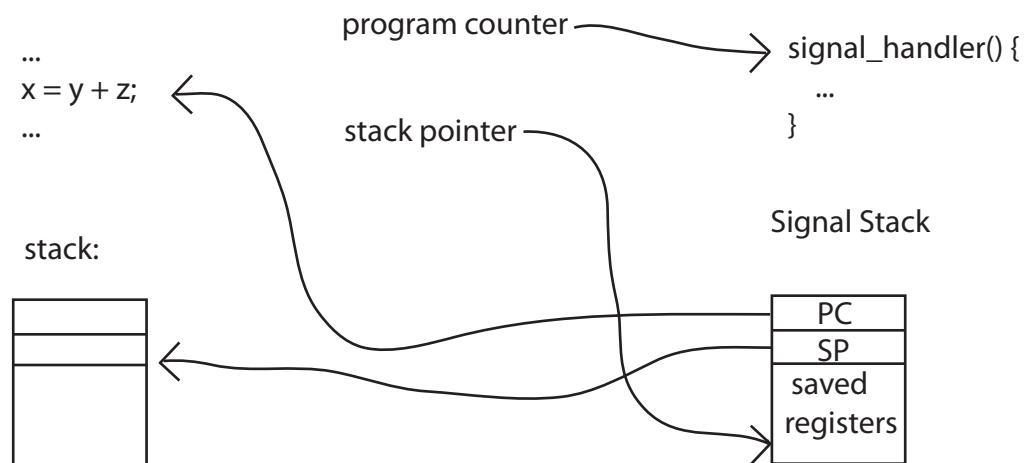


Figure 2.16: The state of the user program and signal handler during a UNIX signal. UNIX signals behave analogously to hardware exceptions, but at user-level.

Exercises

5. Define three styles of switching from user-mode to kernel-mode, and four styles of switching from kernel-mode to user-mode.
6. A typical hardware architecture provides an instruction called return from interrupt, abbreviated by something like iret. This instruction switches the mode of operation from kernel-mode to user-mode. This instruction is usually only available while the machine is running in kernel-mode.
 - a. Explain where in the operating system this instruction would be used.
 - b. Explain what happens if an application program executes this instruction.
7. A hardware designer argues that there are enough transistors on the chip to provide 1024 integer registers and 512 floating point registers, so that the compiler almost never needs to store anything on the stack. You have been invited as the operating system guru to give an opinion on the new design.
 - a. What is the effect of having such a large number of registers on the operating system?
 - b. What additional hardware features you would recommend adding to the design above?
 - c. What happens if the hardware designer also wants to add a 16-station pipeline into the CPU, with precise exceptions. How would that affect the user-kernel switching overhead?

2.4 Case Study: Booting an operating system kernel

When a computer initially starts, it sets the machine's program counter to start executing at a pre-determined position in memory. As the computer has not started running at this point, the initial machine instructions must be ready to be fetched and executed immediately when the power is turned on. For this, systems typically use a special read-only hardware memory (*Boot ROM*) to store these boot instructions. On most x86 personal computers, the boot program is called the BIOS, for "Basic Input/Output System".

Definition: **Boot ROM**

What does the BIOS need to do? We could try to store the machine instructions for the entire operating system in ROM, but this has several drawbacks.

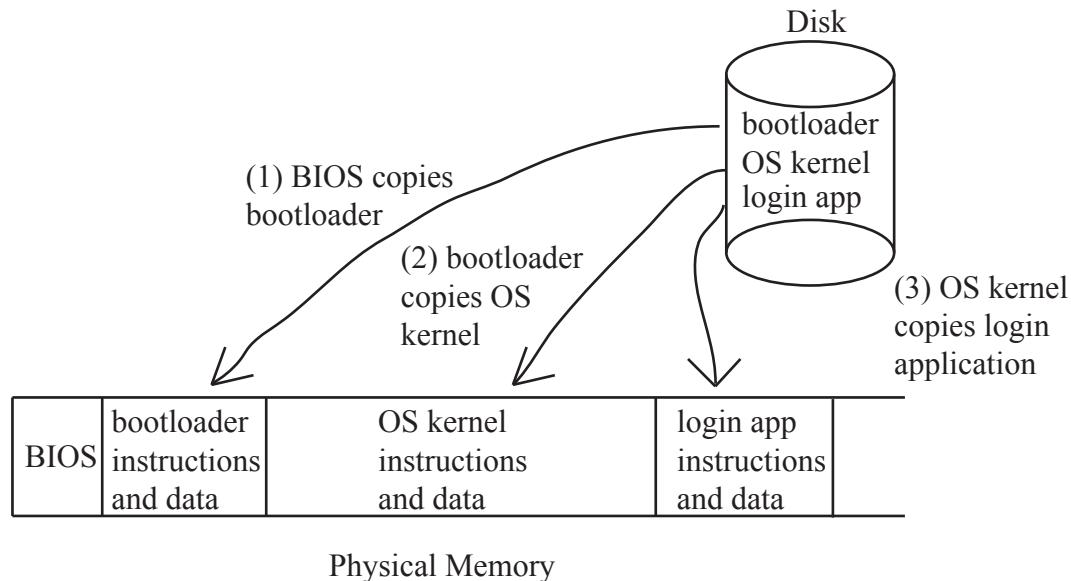


Figure 2.17: The boot ROM copies the bootloader image from disk into memory, and the bootloader copies the operating system kernel image from disk into memory.

The most significant problem is that this would make the operating system difficult to update. The instructions stored in ROM are fixed at the time the computer is manufactured and (except in rare cases) never changed. If an error occurs while the BIOS is being updated, the machine can be left in a permanently unusable state — unable to boot and unable to complete the update of the BIOS.

By contrast, operating systems are updated quite frequently, as bugs and security vulnerabilities are discovered and fixed. ROM storage is also relatively slow and expensive.

Instead, the BIOS provides a level of indirection, as illustrated in Figure 2.17. The BIOS reads a fixed size block of bytes from a fixed position on disk (or flash RAM) into a fixed position in memory. This block of bytes is called the *bootloader*. Once the BIOS has copied the bootloader into memory, it then jumps to the first instruction in the block. On some newer machines, the BIOS also checks that the bootloader has not been corrupted by a virus. (Needless to say, if a virus is able to overwrite the bootloader and get the BIOS to jump to it, the virus can then do whatever it wants with the machine.) As a check, the bootloader is stored with a cryptographic signature. A *cryptographic signature*

Definition: **bootloader**

Definition: **cryptographic signature**

is a hash function of the bytes in a file, such that it is computationally intractable for an attacker to create a different file that matches the signature of the true file. The BIOS checks the signature before jumping to the code, verifying its authenticity.

The bootloader in turn loads the actual operating system kernel into memory, and jumps to it. Again, the bootloader can check the cryptographic signature of the operating system to verify that it has not been corrupted by a virus. The operating system kernel executable image is usually stored in the file system. Thus, to find the bootloader, the BIOS needs to know how to read a block of raw bytes from disk. The bootloader needs to know how to read bytes from the file system to find and read the operating system image.

When the operating system kernel starts running, it can initialize its data structures, including setting up the interrupt table to point to the various interrupt, exception and system call handlers. The kernel then starts the first process, typically the user login page. To run this process, the operating system reads the code for the login program from where it is stored on disk, and jumps to the first instruction in the program, using the start process procedure described above to safely transition control to user-level. The login process in turn can trap into the kernel using a system call whenever it needs the kernel's services, e.g., to render the login prompt on the screen. We will discuss what system calls are needed for processes to do useful work in the next chapter.

2.5 Case Study: Virtual machines

Some operating system kernels provide the abstraction of a entire virtual machine at user-level. How do interrupts, exceptions, and system calls work in this context? To avoid confusion when discussing virtual machines, we need to remind you of some terminology we introduced in Chapter 1. The operating system providing the virtual machine abstraction is called the *host operating system*. The operating system running inside the virtual machine is called the *guest operating system*.

Definition: host operating system

Definition: guest operating system

The guest operating system needs to be able to do everything a real operating system would do. For example, to provide a guest disk, the host operating system simulates a virtual disk as a file on the physical disk. To provide network access to the guest operating system, the host operating system simulates a virtual network using physical network packets. Likewise, the host operating system needs to manage memory to provide the illusion that the guest operating system is managing its own memory protection, even though it is running with virtual addresses. We will discuss address translation for virtual machines in more detail in a later chapter.

Here we focus on how the host operating system manages the control transfer between processes running on the guest operating system and the guest OS itself.

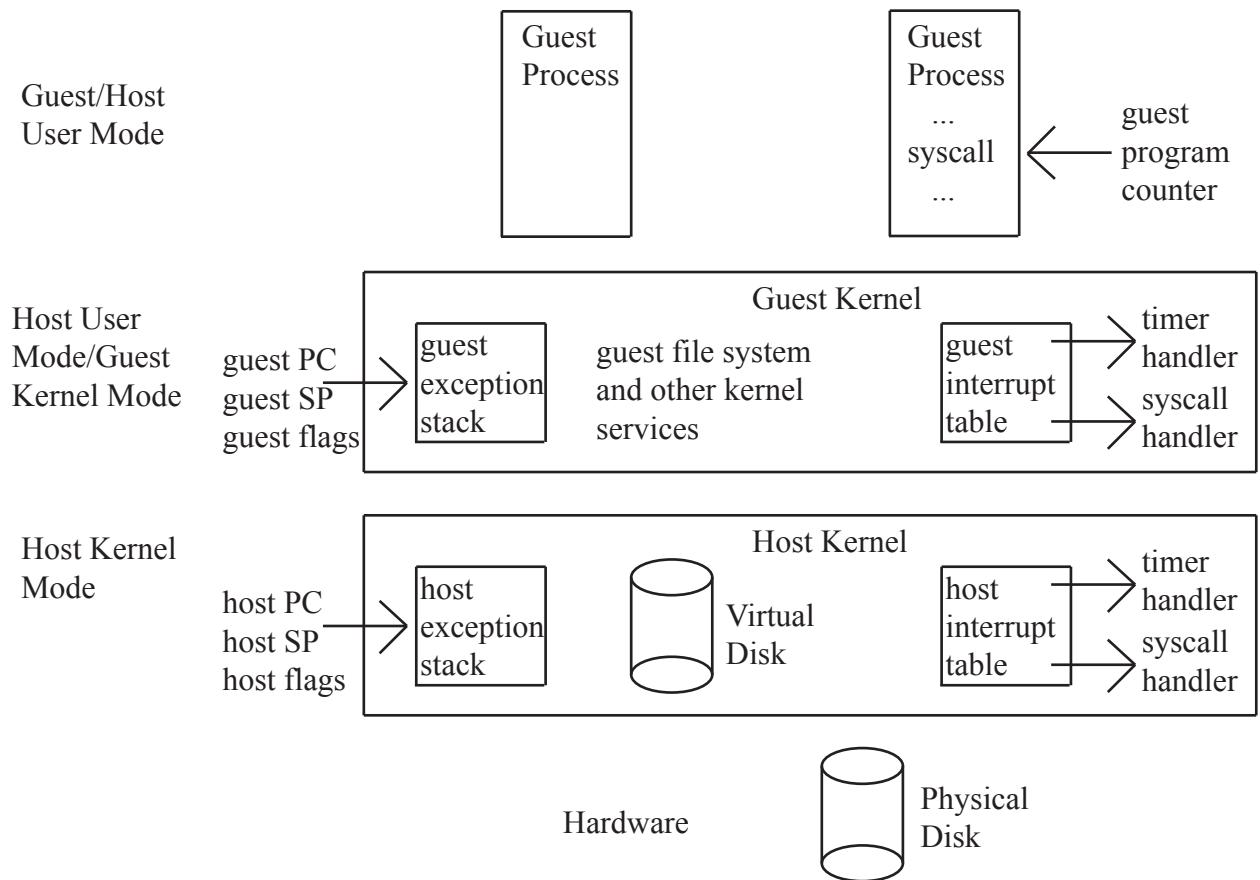


Figure 2.18: Emulation of user/kernel mode transfer for processes running inside a virtual machine.

During boot, the host operating system initializes its interrupt table to point to its own interrupt handlers in the host kernel memory region. When the host operating system starts the virtual machine, the guest operating system starts running as if it is being booted:

- The host operating system loads the guest bootloader from the virtual disk and starts it running.
- The guest bootloader loads the guest operating system from the virtual disk into memory and starts it running.
- The guest operating system then initializes its own interrupt tables to point to the interrupt handlers within the guest kernel.

- The guest operating system loads a process from the virtual disk into memory.
- To start a process, the guest operating system issues instructions to resume execution at user-level, e.g., using `reti` on the x86. As changing the privilege level is a privileged operation, this instruction will trap into the host operating system kernel.
- The host operating system simulates the requested mode switch as if the processor had directly executed it, restoring the program counter, stack pointer, and processor status word exactly as the guest operating system had intended. Note that the host operating system needs to protect itself from bugs in the guest operating system, and so it also needs to check the validity of the mode switch — e.g., that the guest operating system is not surreptitiously attempting to get the host kernel to “switch” to an arbitrary point in the kernel code.

Next, consider what happens when the guest user process does a system call, illustrated in Figure 2.18. To the hardware, there is only one kernel, the host operating system. Thus, the trap instruction will trap into the host kernel’s system call handler. Of course, the system call was not for the host! Rather, the host kernel simulates what would have happened had the system call instruction occurred on real hardware running the guest operating system:

- The host kernel saves the instruction counter, processor status register, and user stack pointer in the exception stack of the guest operating system.
- The host kernel transfers control to the guest kernel at the beginning of the interrupt handler, but with the guest kernel running with user-mode privilege.
- The guest kernel performs the system call.
- When the guest kernel attempts to return from the system call back to user-level, this will cause a privilege exception, dropping back into the host operating system kernel.
- The host kernel can then restore the state of the user process, running at user level, as if the guest operating system had been able to return there directly.

Exceptions are handled similarly, with one caveat. Some exceptions generated by the virtual machine are due to the user process; these are forwarded to the guest kernel for handling. Other exceptions are generated by the guest kernel itself (e.g., when it tries to execute privileged instructions); these must be handled by the host kernel. Thus, the host kernel needs to keep track of

whether the virtual machine is executing in virtual user mode or virtual kernel mode.

Hardware interrupts are vectored by the hardware to the host kernel. Special handling is needed for time: time can elapse in the host without elapsing in the guest. When a timer interrupt occurs, it may be that enough virtual time has passed that the guest kernel is due a timer interrupt; in that case, the host kernel returns from the interrupt to the interrupt handler for the guest kernel. The guest kernel may in turn switch guest processes; its return from interrupt will cause a privilege exception, returning back to the host kernel, which can then resume the correct guest process.

Handling input/output interrupts is even simpler, as the simulation of the virtual device does not need to be anything like a real device. When the guest kernel makes a request to a virtual disk, it writes instructions to the buffer descriptor ring for the disk device; the host kernel will need to translate these instructions into operations on the virtual disk. The host kernel can simulate the disk request however it likes — e.g., through regular file reads and writes, copied into the guest kernel memory as if there was true DMA hardware. The guest kernel will expect to receive an interrupt when the virtual disk completes its work; this can be triggered by the timer interrupt as described above, but vectored to the guest disk interrupt handler instead of the guest timer interrupt handler.

2.6 Conclusion and future directions

The process concept – the ability to execute arbitrary user programs with restricted rights – has been remarkably successful. With the exception of devices that run only a single application at a time (such as embedded systems and game consoles), every commercially successful operating system started in the past two decades has provided process isolation and several existing systems have switched over.

The reason for this success is obvious. Without process isolation, computer systems would be much more fragile and less secure. As recently as a decade ago, it was common for personal computers to crash on a daily basis. Today, it is not unusual for laptops to remain working for weeks at a time without rebooting. This has occurred even though the operating system and application software on these systems has become more complex. While some of the improvement is due to factors such as better hardware reliability and automated bug tracking, process isolation has been a key technology in constructing more reliable systems software.

Process isolation is also essential to building more secure computer systems. Without isolation, computer users would be forced to trust everything loaded onto the computer — not just the operating system code, but every application

Hardware support for operating systems

In this chapter, we have described a number of hardware mechanisms to support operating systems:

- Privilege levels: user and kernel
- Privileged instructions: instructions available only in kernel mode
- Memory translation: to prevent user programs from accessing kernel data structures, and to aid in memory management
- Exceptions: trap to the kernel on a privilege violation or other unexpected event
- Timer interrupts: return control to the kernel on time expiration
- Device interrupts: return control to the kernel to signal I/O completion
- Interprocessor interrupts: cause another processor to return control to the kernel
- System calls: trap to the kernel to perform a privileged action on behalf of a user program
- Return from interrupt: switch from kernel-mode to user-mode, to a specific location in a user program
- Boot ROM: fixed code to load startup routines from disk into memory

To support threads, we will need one additional mechanism, described in a later chapter:

- Atomic instructions: instructions to atomically read and modify a memory location, used to implement synchronization in multithreaded programs
-
-

installed on the system. In practice, however, complete process isolation is still more of an aspiration than a reality. Most operating systems are vulnerable to malicious applications, because the attacker can exploit any vulnerability in the implementation. Although keeping your system up to date with the latest patches provides some level of defense, it is still inadvisable to download and install untrusted software off the web.

In the future, we are likely to see three complementary trends:

- **Operating system support for fine-grained protection.** Process isolation is evolving to be more flexible and fine-grained, to reflect different levels of trust in different applications. Even if the operating system is successfully hardened against rogue applications, it is typical for an application invoked by a user to have the permissions of that user. In other words, a virus masquerading as a screen saver does not need to compromise the operating system to steal or corrupt that user's data. Smartphone operating systems have been the first to add these types of

controls — to prevent certain applications without a “need to know” from accessing sensitive information, such as the smartphone’s location or the list of frequently called telephone numbers.

- **Operating system support for application-layer sandboxing.** Increasingly, many applications are becoming mini-operating systems in their own right, capable of safely executing third party software to extend and improve the user experience. Scripts embedded in web pages have become increasingly sophisticated; web browsers need to be able to efficiently and completely isolate these scripts so that they cannot steal the user’s data or corrupt the browser. Other applications such as databases and desktop publishing systems are also moving in the direction of needing application-layer sandboxing. Google’s NativeClient and Microsoft’s AppDomains are two example systems that provide general-purpose safe execution of third party code at the user-level.
- **Hardware support for virtualization.** Virtual machines provide an extra layer of protection beneath the operating system. Even if a malicious process run by a guest operating system on a virtual machine is able to corrupt the kernel, its impact will be limited to just that virtual machine. Below the virtual machine interface, the host operating system needs to provide isolation between different virtual machines; this is much easier in practice because the virtual machine interface is much simpler than the operating system kernel’s system call interface. For example, in a data center, virtual machines provide users the flexibility to run any application, without compromising the data center operation.

For this to be practical, processor architectures are being re-designed to reduce the cost of running a virtual machine. For example, on some new processors, guest operating systems can directly handle their own system calls, interrupts and exceptions, without those events needing to be mediated by the host operating system implementing the virtual machine. Likewise, I/O devices are being re-designed to do direct transfers to and from the guest operating system, without the need to go through the host kernel.

Exercises

For convenience, the exercises from the body of the chapter are repeated here.

1. We mentioned that for the “Hello world” program, the kernel must copy the string from the user program into the screen memory. Why must the screen’s buffer memory be protected? Explain what might happen if a (malicious) application could alter any pixel on the screen, not just those within its own window.

2. For each of the three mechanisms for supporting dual mode operation — privileged instructions, memory protection, and timer interrupts — explain what might go wrong without that mechanism, assuming the system still had the other two.
3. Suppose we had a perfect object-oriented language and compiler, so that only an object's methods could access the internal data inside an object. If the operating system only ran programs written in that language, would it still need hardware memory address protection?
4. Suppose you are tasked with designing the security system for a new web browser that supports rendering web pages with embedded web page scripts. What checks would you need to ensure that executing buggy or malicious scripts could not corrupt or crash the browser?
5. Define three styles of switching from user-mode to kernel-mode, and four styles of switching from kernel-mode to user-mode.
6. A typical hardware architecture provides an instruction called return from interrupt, abbreviated by something like iret. This instruction switches the mode of operation from kernel-mode to user-mode. This instruction is usually only available while the machine is running in kernel-mode.
 - a. Explain where in the operating system this instruction would be used.
 - b. Explain what happens if an application program executes this instruction.
7. A hardware designer argues that there are enough transistors on the chip to provide 1024 integer registers and 512 floating point registers, so that the compiler almost never needs to store anything on the stack. You have been invited as the operating system guru to give an opinion on the new design.
 - a. What is the effect of having such a large number of registers on the operating system?
 - b. What additional hardware features you would recommend adding to the design above?
 - c. What happens if the hardware designer also wants to add a 16-station pipeline into the CPU, with precise exceptions. How would that affect the user-kernel switching overhead?
8. Which of the following components is responsible for loading the initial value in the program counter for an application program before it starts running: the compiler, the linker, the kernel, or the boot ROM?

9. We described how the operating system kernel mediates access to I/O devices for safety. Some newer I/O devices are *virtualizable* — they permit safe access from user-level programs, such as a guest operating system running in a virtual machine. Explain how you might design the hardware and software to get this to work. (Hint: For this, the device needs much of the same hardware support as the operating system kernel.)
10. System Calls vs. Procedure Calls: How much more expensive is a system call than a procedure call? Write a simple test program to compare the cost of a simple procedure call to a simple system call (`getpid()` is a good candidate on UNIX; see the man page.) To prevent the optimizing compiler from “optimizing out” your procedure calls, do not compile with optimization on. You should use a system call such as the UNIX `gettimeofday()` for time measurements. Design your code such that the measurement overhead is negligible. Also, be aware that timer values in some systems have limited resolution (e.g., millisecond resolution).
Explain the difference (if any) between the time required by your simple procedure call and simple system call by discussing what work each call must do.
11. Suppose you have to implement an operating system on hardware that supports interrupts and exceptions but that does not have a trap instruction. Can you devise a satisfactory substitute for traps using interrupts and/or exceptions? If so, explain how. If not, explain why.
12. Suppose you have to implement an operating system on hardware that supports exceptions and traps but that does not have interrupts. Can you devise a satisfactory substitute for interrupts using exceptions and/or traps? If so, explain how. If not, explain why.
13. Explain the steps that an operating system goes through when the CPU receives an interrupt.
14. When an operating system receives a system call from a program, a switch to the operating system code occurs with the help of the hardware. In such a switch, the hardware sets the mode of operation to kernel-mode, calls the operating system trap handler at a location specified by the operating system, and allows the operating system to return back to user mode after it finishes its trap handling.

Consider the stack on which the operating system must run when it receives the system call. Should this be a different stack from the one that the application uses, or could it use the same stack as the application program? Assume that the application program is blocked while the system call runs.

15. Write a program to verify that the operating system on your computer protects itself from rogue system calls correctly. For a single system call such as file system open, try all possible illegal calls: e.g., an invalid system call number, an invalid stack pointer, an invalid pointer stored on the stack, etc. What happens?

Chapter 3

The Programming Interface

From a programmer's point of view, the user is a peripheral that types when you issue a read request. – Peter Williams

The previous chapter concerned the mechanisms needed in the operating system kernel to implement the process abstraction. A process is an instance of a program — the kernel provides an efficient sandbox for executing untrusted code at user-level, running user code directly on the processor.

This chapter concerns how we choose to use the process abstraction: what functionality does the operating system provide applications, and what should go where — what functionality should be put in the operating system kernel, what should be put into user-level libraries, and how should the operating system itself be organized?

There are as many answers to this as there are operating systems. Describing the full programming interface and internal organization for even a single operating system would take an entire book. Instead, in this chapter we explore a subset of the programming interface for UNIX, the foundation of Linux, Mac OS, iOS, and Android. We also touch on how the same issues are addressed in Windows.

First, we need to answer “what” — what functions do we need an operating system to provide applications?

- **Process management.** Can a program create an instance of another program? Wait for it to complete? Stop or resume another running program? Send it an asynchronous event?
- **Input/output.** How do processes communicate with devices attached to the computer and through them to the physical world? Can processes communicate with each other?

- **Thread management.** Can we create multiple activities or threads that share memory or other resources within a process? Can we stop and start threads? How do we synchronize their use of shared data structures?
- **Memory management.** Can a process ask for more (or less) memory space? Can it share the same physical memory region with other processes?
- **File systems and storage.** How does a process store the user's data persistently so that it can survive machine crashes and disk failures? How does the user name and organize their data?
- **Networking and distributed systems.** How do processes communicate with processes on other computers? How do processes on different computers coordinate their actions despite machine crashes and network problems?
- **Graphics and window management.** How does a process control pixels on its portion of the screen? How does a process make use of graphics accelerators?
- **Authentication and security.** What permissions does a user or a program have, and how are these permissions kept up to date? On what basis do we know the user (or program) is who they say they are?

In this chapter, we focus on just the first two of these topics: process management and input/output. We will cover thread management, memory management, and file systems in detail in later chapters in this book. We expect to add chapters on networks, distributed systems and security in later releases of this book. Typically, the graphics systems is covered in a specialized course on that topic, and so we do not plan to include it here.

Remarkably, we can describe a functional interface for process management and input/output with just a dozen system calls, and the rest of the system call interface with another dozen. Even more remarkably, these calls are nearly unchanged from the original UNIX design. Despite being first designed and implemented in 1973, most of these calls are still in wide use in systems today!

Second, we need to answer “where” — for any bit of functionality the operating system provides to user programs, we have several options for where it lives, illustrated in Figure 3.1:

- We can put the functionality in a user-level program. In both Windows and UNIX, for example, there is a user program for managing a user's login and another for managing a user's processes.
- We can put the functionality in a user-level library linked in with each application. In Windows and Mac OS, user interface widgets are part of user-level libraries, included in those applications that need them.

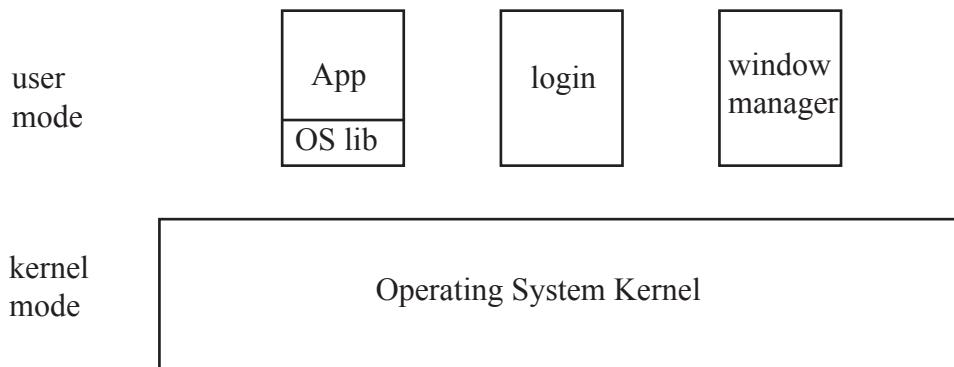


Figure 3.1: Operating system functionality can be implemented in user-level programs, in user-level libraries, in the kernel itself, or in a user-level server invoked by the kernel.

- We can put the functionality in the operating system kernel, accessed through a system call. In Windows and UNIX, low level process management, the file system and the network stack are all implemented in the kernel.
- We can access the function through a system call, but implement the function in a standalone server process invoked by the kernel. In many systems, the window manager is implemented as a separate server process.

How do we make this choice? It is important to realize that the choice can be (mostly) transparent to both the user and the application programmer. The user wants a system that works; the programmer wants a clean, convenient interface that does the job. As long as the operating system provides that interface, where each function is implemented is up to the operating system, based on a tradeoff between flexibility, reliability, performance and safety.

- **Flexibility.** It is much easier to change operating system code that lives outside of the kernel, without breaking applications using the old interface. If we create a new version of a library, we can just link that library in with new applications, and over time convert old applications to use the new interface. However, if we need to change the system call interface, we must either simultaneously change both the kernel and all applications, or we must continue to support both the old and the new versions until all old applications have been converted. Since many applications may be written by third party developers, outside of the control of the operating

system vendor, changing the system call interface is a huge step, often requiring coordination across many companies.

One of the key ideas in UNIX, responsible for much of its success, was to design its system call interface to be simple and powerful, so that almost all of the innovation in the system could happen in user code without changing the interface to the operating system. The UNIX system call interface is also highly portable — the operating system could be ported to new hardware without needing to rewrite application code. As shown in Figure 3.2, the kernel can be seen as a “thin waist”, enabling innovation at the application-level, and in the hardware, without requiring simultaneous changes in the other parts of the system.

- **Safety.** However, resource management and protection must be implemented in the operating system kernel, or in a specially privileged process called by the kernel. As we explained in the previous chapter, if applications can directly execute instructions on the processor, they can skip any protection code in a user-level library, so protection checks cannot be implemented at that level.
- **Reliability.** Improved reliability is another reason to keep the operating system kernel minimal. Kernel code needs the power to set up hardware devices, such as the disk, and to control protection boundaries between applications. However, kernel modules are typically not protected from one another, and so a bug in kernel code (whether sensitive or not) may corrupt user or kernel data. This has led some systems to use a philosophy of “what can be at user level, should be.” An extreme version of approach is to isolate privileged, but less critical, parts of the operating system such as the file system or the window system, from the rest of the kernel. This is called a *microkernel* design. In a microkernel, the kernel itself is kept small, and instead most of the functionality of a traditional operating system kernel is put into a set of user-level processes, or servers, accessed from user applications via interprocess communication.
- **Performance.** Finally, transferring control into the kernel is more expensive than a procedure call to a library, and transferring control to a user-level file system server via the kernel is still even more costly. Modern processor hardware has added various support to reduce the cost of these boundary crossings, but the performance issue remains important. Microsoft Windows NT, a precursor to Windows 7, was initially designed as a microkernel, but over time much of its functionality has been migrated back into the kernel for performance reasons.

There are no easy answers! We will investigate the question of how to design the system call interface and where to place operating system functionality through case studies of UNIX and other systems:

Definition: **microkernel**

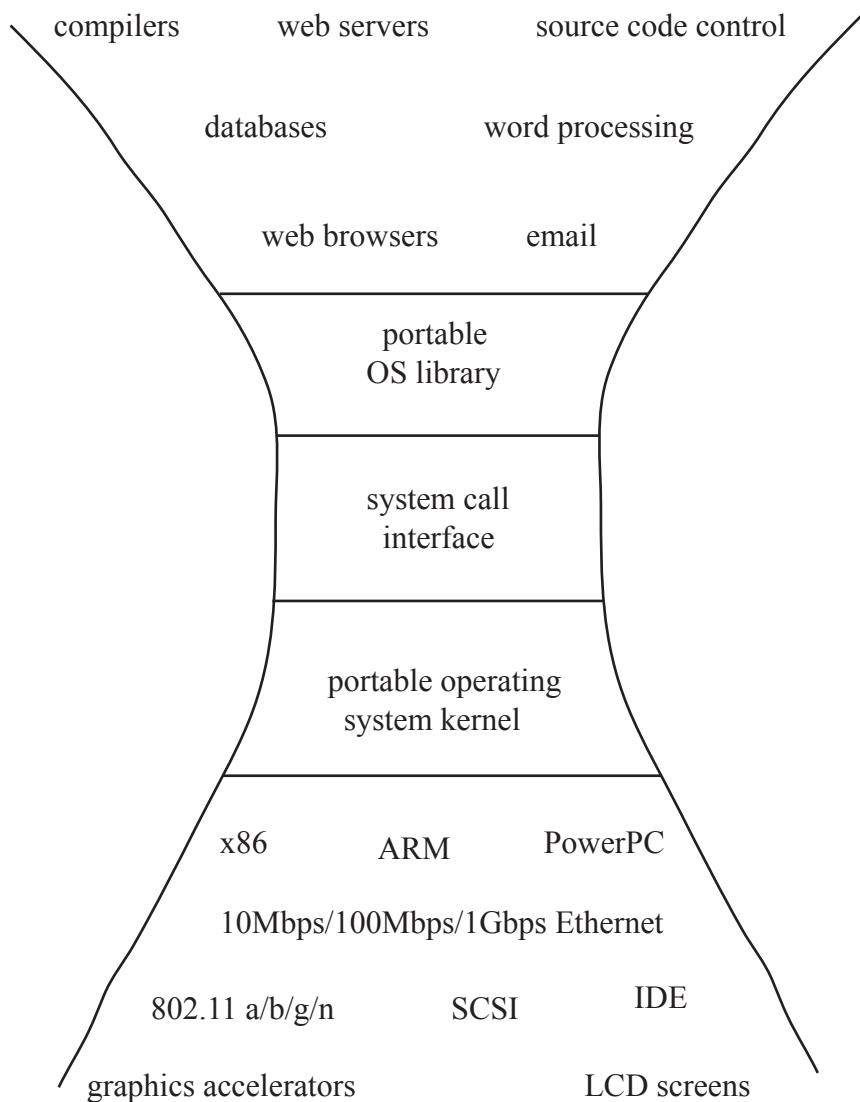


Figure 3.2: The kernel system call interface can be seen as a “thin waist”, enabling independent evolution of applications and hardware.

The Internet and the “thin waist”

The Internet is another example of the benefit of designing interfaces to be simple and portable. The Internet defines a packet-level protocol that can run on top of virtually any type of network hardware and can support almost any type of network application. Creating the World Wide Web required no changes to the Internet packet delivery mechanism; likewise, the introduction of wireless networks required changes in hardware devices and in the operating system, but no changes in network applications. Although the Internet’s “thin waist” can sometimes lead to inefficiencies, the upside is to foster innovation in both applications and hardware by decoupling changes in one from changes in the other.

Application-level sandboxing and operating system functionality

Applications that support executing third-party code or scripts in a restricted sandbox must address many of these same questions, with the sandbox playing the role of the operating system kernel. In terms of functionality: Can the scripting code start a new instance of itself? Can it do input/output? Can it perform work in the background? Can it store data persistently, and if it can, how does it name that data? Can it communicate data over the network? How does it authenticate actions?

For example, in web browsers, HTML5 not only allows scripts to draw on the screen, communicate with servers, and save and read cookies, it also has recently added programming interfaces for offline storage and cross-document communication. The Flash media player provides scripts with the ability to do asynchronous operations, file storage, network communication, memory management, and authentication.

Just as with system calls, these interfaces must be carefully designed to be bullet-proof against malicious use. A decade ago, email viruses became widespread because scripts could be embedded in documents that were executed on opening; the programming interfaces for these scripts would allow them to discover the list of correspondents known to the current email user and to send them email, thereby propagating and expanding the virus with a single click. The more fully featured the interface, the more convenient it is for developers, and the more likely that some aspect of the interface will be abused by a hacker.

- **Process management.** What is the system call interface for process management?
- **Input/output.** What is the system call interface for performing I/O and interprocess communication?
- **Case study: Implementing a shell.** We will illustrate these interfaces by using them to implement a user-level job control system called a *shell*.
- **Case study: Interprocess communication.** How does the communication between a client and server work?
- **Operating system structure.** Can we use the process abstraction to simplify the construction of the operating system itself and to make it more secure, more reliable, and more flexible?

3.1 Process management

On a modern computer, when a user clicks on a file or application icon, the application starts up. How does this happen and who gets called? Of course, we could implement everything that needs to happen in the kernel — draw the icon for every item in the file system, map mouse positions to the intended icon, catch the mouse click, and start the process. In early batch processing systems, the kernel was in control by necessity. Users submitted jobs, and the operating system took it from there, instantiating the process when it was time to run the job.

A different approach is to allow user programs to create and manage their own processes. This has fostered a blizzard of innovation. Today, programs that create and manage processes include window managers, web servers, web browsers, shell command line interpreters, source code control systems, databases, compilers, and document preparation systems. We could go on, but you get the idea. If creating a process is something a process can do, then anyone build a new version of any of these applications, without recompiling the kernel or forcing anyone else to use it.

An early motivation for user-level process management was to allow developers to write their own shell command line interpreters. A *shell* is a job control system; both Windows and UNIX have a shell. Many tasks involve a sequence of steps to do something, each of which can be its own program. With a shell, you can write down the sequence of steps, as a sequence of programs to run to do each step. Thus, you can view it as a very early version of a scripting system.

Definition: **shell**

For example, to compile a C program from multiple source files, you might type:

If we put those commands into a file, the shell reads the file and executes it, creating, in turn, a process to compile sourcefile1.c, a process to compile

```
# this program compiles sourcefile1.c and sourcefile2.c
# and then links the results together into an executable program
cc -c sourcefile1.c
cc -c sourcefile2.c
ln -o program sourcefile1.o sourcefile2.o
```

There is an app for that

User-level process management is another way of saying “there is an app for that.” Instead of a single program that does everything, we can create specialized programs for each task, and mix and match what we need. The formatting system for this textbook uses over fifty separate programs.

The web is a good example of the power of composing complex applications from more specialized services. A web page does not need to do everything itself: it can mash up the results of many different web pages, and it can invoke process creation on the local server to generate part of the page. The flexibility to create processes was extremely important early on in the development of the web. HTML was initially just a way to describe the formatting for static information, but it included a way to escape to a process to, say, do a lookup in a database or to authenticate a user. Over time, HTML has added support for many different features that were first prototyped via execution by a separate process. And of course, HTML still retains the ability to execute a process for everything not supported by the standard.

sourcefile2, and a process to link them together. Once a shell script is a program, we can create other programs by combining scripts together. In fact, on UNIX, the C compiler is itself a shell program! The compiler first invokes a process to expand header include files, then a separate process to parse the output, another process to generate (text) assembly code, and yet another to convert assembly into executable machine instructions.

3.1.1 Windows process management

One approach to process management is to just add a system call to create a process, and other system calls for other process operations. This turns out to be simple in theory and complex in practice. In Windows, there is a routine called, unsurprisingly, `CreateProcess`, in simplified form below:

Definition: **parent** We call the process creator the *parent* and the process being created the
Definition: **child** *child*.

What steps does `CreateProcess` take? As we explained in the previous chapter, the kernel needs to:

- Create and initialize the process control block (PCB) in the kernel

```
// simplified version of the call to create a process with arguments on Windows
simplified: boolean CreateProcess(char *prog, char *args)

// Start the child process
if( !CreateProcess( NULL,      // No module name (use command line)
                    argv[1],    // Command line
                    NULL,        // Process handle not inheritable
                    NULL,        // Thread handle not inheritable
                    FALSE,       // Set handle inheritance to FALSE
                    0,           // No creation flags
                    NULL,        // Use parent's environment block
                    NULL,        // Use parent's starting directory
                    &si,         // Pointer to STARTUPINFO structure
                    &pi )        // Pointer to PROCESS_INFORMATION structure
)
```

Figure 3.3: Excerpt from an example of how to use the Windows CreateProcess system call. The first two arguments specify the program and its arguments; the rest concern aspects of the process runtime environment.

- Create and initialize a new address space
- Load the program `prog` into the address space
- Copy arguments `args` into memory in the address space
- Initialize the hardware context to start execution at “start”
- Inform the scheduler that the new process is ready to run

Unfortunately, there are quite a few aspects of the process that the parent might like to control, such as: its privileges, where it sends its input and output, what it should store its files, what to use as a scheduling priority, and so forth. We can’t trust the child process itself to set its own privileges and priority, and it would be inconvenient to expect every application to include code for figuring out its context. So the real interface to `CreateProcess` is quite a bit more complicated in practice, given in Figure 3.3.

3.1.2 UNIX process management

UNIX takes a different approach to process management, one that is complex in theory and simple in practice. UNIX splits `CreateProcess` in two steps, called `fork` and `exec`, illustrated in Figure 3.4.

UNIX `fork` creates a complete copy of the parent process, with one key exception described below. This copy can then do whatever is necessary to set up the context of the child. Because the child process is still running the code of the parent, the copy can be trusted to set up privileges and priorities correctly. Once the context is set, the copy then calls UNIX `exec` to copy in the

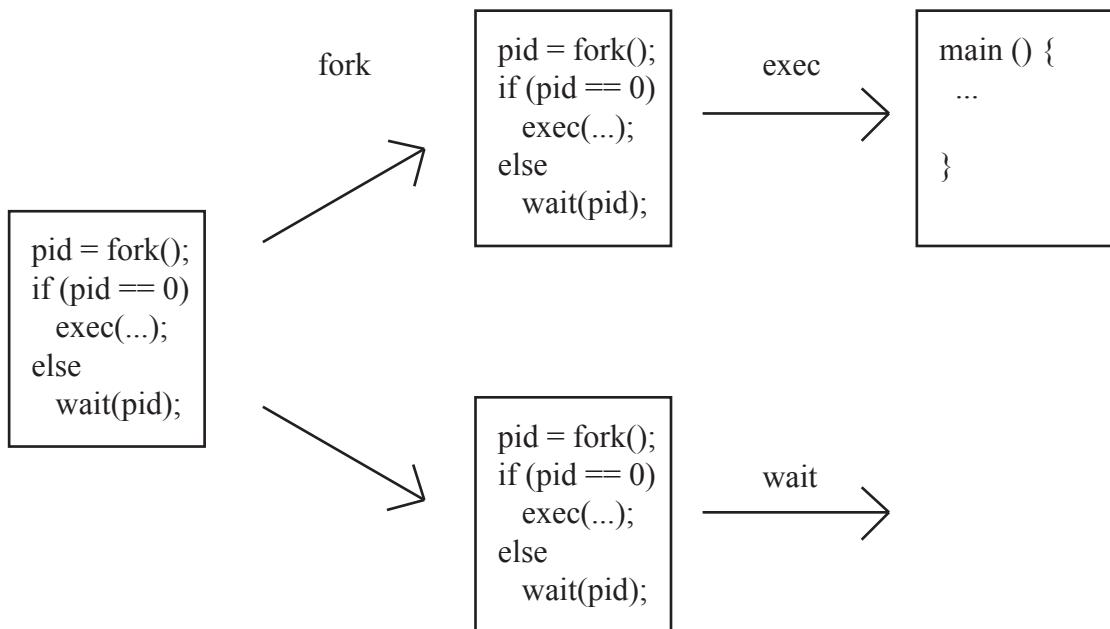


Figure 3.4: The operation of the UNIX fork and exec system calls.

new program and start running it. Although it may seem incredibly wasteful to make a complete copy of a program just to throw it away during `exec`, in a later chapter we will describe a set of techniques that allow UNIX `fork` and `exec` to be implemented with very little copying and a modest amount of bookkeeping by the kernel.

With this design, UNIX `fork` takes no arguments and returns an integer. UNIX `exec` takes only two arguments (the name of the program to run and an array of arguments to pass to the program). In part because of its simplicity, this interface has remained nearly unchanged since UNIX was designed in the early 70's. (Although the interface hasn't changed, the word `fork` is now a bit ambiguous. It is used for creating a new copy of a UNIX process, and in thread systems for creating a new thread. To disambiguate, we will always use the term "UNIX fork" to refer to UNIX's copy process system call.)

UNIX fork

The steps for implementing UNIX `fork` in the kernel are:

- Create and initialize the process control block (PCB) in the kernel
- Create a new address space

```

int child_pid = fork();
if (child_pid == 0) {                                // I'm the child process
    // getpid is a system call to get the current process's ID
    printf("I am process #%d\n", getpid());
    return 0;
} else {                                         // I'm the parent
    printf("I am parent of process #%d\n", child_pid);
    return 0;
}

Possible output:
I am parent of process 495
I am process 495

Another less likely but still possible output:
I am process 456
I am parent of process 456

```

Figure 3.5: Example UNIX code to fork a process, and some possible outputs of running the code.

- Initialize the address space with a copy of the entire contents of the address space of the parent
- Inherit the execution context of the parent (e.g., any open files)
- Inform the scheduler that the new process is ready to run

A strange aspect of UNIX `fork` is that the system call returns *twice*: once to the parent and once to the child. To the parent, UNIX returns the process ID of the child; to the child, it returns 0 indicating success. Just as if you made a clone of yourself, you would need some way to tell who was the clone and who was the original, UNIX uses the return value from `fork` to distinguish the two copies. Some sample code to call `fork` is given in Figure 3.5.

If we run the program in Figure 3.5, what happens? If you have access to a UNIX system, you can try it and see for yourself. UNIX `fork` returns twice, once in the child, with a return value of 0, and once in the parent with a return value of the child's process ID. However, we do not know whether the parent will run next or the child. The parent had been running, and so it is likely that it will reach its print statement first. However, a timer interrupt could intervene between when the parent forks the process and when it reaches the print statement, so that the processor is reassigned to the child. Or we could be running on a multicore, where both the parent and child are running simultaneously. In either case, the child could print its output before the parent. We will talk in much more depth about the implications of different orderings of concurrent execution in the next chapter.

UNIX fork and the Chrome Web browser

Although UNIX `fork` is normally paired with a call to `exec`, in some cases UNIX `fork` is useful on its own. A particularly interesting example is in Google's Chrome web browser. When the user clicks on a link, Chrome forks a process to fetch and render the web page at the link, in a new tab on the browser. The parent process continues to display the original referring web page, while the child process runs the same browser, but in its own address space and protection boundary. The motivation for this design is to isolate the new link, so that if the web site is infected with a virus, it won't infect the rest of the browser. Closing the infected browser tab will then remove the link and the virus from the system.

Some security researchers take this a step further. They set up their browsers and email systems to create a new *virtual machine* for every new link, running a copy of the browser in each virtual machine; even if the web site has a virus that corrupts the guest operating system running in the virtual machine, the rest of the system will remain unaffected. In this case, closing the virtual machine cleans the system of the virus.

Interestingly, on Windows, Google Chrome does not use `CreateProcess` to fork new copies of the browser on demand. The difficulty is that if Chrome is updated while Chrome is running, `CreateProcess` will create a copy of the new version, and that may not interoperate correctly with the old version. Instead, they create a pool of helper processes that wait in the background for new links to render.

UNIX exec and wait

The UNIX system call `exec` completes the steps needed to start running a new program. UNIX `exec` is typically called by the child process after it has returned from UNIX `fork` and configured the execution environment for the child. We will describe more about how this works when we discuss UNIX pipes in the next section.

UNIX `exec` does the following steps:

- Load the program `prog` into the current address space
- Copy arguments `args` into memory in the address space
- Initialize the hardware context to start execution at “start”

Note that `exec` does not create a new process!

On the other side, often the parent process needs to pause until the child process completes, e.g., if the next step depends on the output of the previous step. In the shell example we started the chapter with, we need to wait for the two compilations to finish before it is safe to start the linker.

Kernel handles and garbage collection

As we discussed in the previous chapter, when a UNIX process finishes, it calls the system call `exit`. `Exit` can release various resources associated with the process, such as the user stack, heap, and code segments. It must be careful, however, in how it garbage collects the process control block (PCB). Even though the child process has finished, if it deletes the PCB, then the parent process will be left with a dangling pointer if later on it calls UNIX `wait`. Of course, we don't know for sure if the parent will ever call `wait`, so to be safe, the PCB can only be reclaimed when both the parent and the child have finished or crashed.

Generalizing, both Windows and UNIX have various system calls that return a handle to some kernel object; these handles are used in later calls as an ID. The process ID returned by UNIX `fork` is used in later calls to UNIX `wait`; we'll see below that UNIX `open` returns a file descriptor that is used in other system calls. It is important to realize that these handles are *not* pointers to kernel data structures; otherwise, an erroneous user program could cause havoc in the kernel by making system calls with fake handles. Rather, they are specific to the process and checked for validity on each use.

Further, in both Windows and UNIX, handles are reference counted. Whenever the kernel returns a handle, it bumps a reference counter, and whenever the process releases a handle (or exits) the reference counter is decremented. UNIX `fork` sets the process ID reference count to two, one for the parent and one for the child. The underlying data structure, the PCB, is reclaimed only when the reference count goes to zero, that is, when both the parent and child terminate.

UNIX has a system call, naturally enough called `wait`, that pauses the parent until the child finishes, crashes, or is terminated. Since the parent could have created many child processes, `wait` is parameterized with the process ID of the child. With `wait`, a shell can create a new process to perform some step of its instructions, and then pause for that step to complete before proceeding to the next step. It would be hard to build a usable shell without `wait`.

However, the call to `wait` is optional in UNIX. For example, the Chrome browser does not need to wait for its forked clones to finish. Likewise, most UNIX shells have an option to run operations in the background, signified by appending an ‘&’ to the command line. (As with `fork`, the word `wait` is now a bit ambiguous. It is used for pausing the current UNIX process to wait for another process to complete; it is also used in thread synchronization, for waiting on a condition variable. To disambiguate, we will always use the term “UNIX `wait`” to refer to UNIX’s `wait` system call. Oddly, waiting for a thread to complete is called “thread join”, even though it is most analogous to UNIX `wait`. Windows is simpler, with a single function called “`WaitForSingleObject`” that can wait for process completion, thread completion, or on a condition variable.)

Finally, as we outlined in the previous chapter, UNIX provides a facility for one process to send another an instant notification, or upcall. In UNIX, the

notification is sent by calling `signal`. Signals are used for terminating an application, suspending it temporarily for debugging, resuming after a suspension, timer expiration, and a host of other reasons. In the default case, where the receiving application did not specify a signal handler, the kernel implements a standard one on its behalf.

Exercises

1. Can UNIX fork return an error? Why or why not?

Note: You can answer this question by looking at the manual page for `fork`, but before you do that, think about what the `fork` system call does. If you were designing this call, would you need to allow `fork` to return an error?

Note: A manual page (or “man page”) is a standard way of documenting Unix system calls and utility programs. On a Unix machine, you can access a manual page by running the `man` command (e.g., `man fork`). Another way to find manual pages is via web search (e.g., search for `man fork` and many of the top results will be manual pages for the `fork` system call.)

2. Can UNIX exec return an error? Why or why not?

Note: You can answer this question by looking at the manual page for `exec`, but before you do that, think about what the `exec` system call does. If you were designing this call, would you need to allow it to return an error?

3. What happens if we run the following program on UNIX?

```
main() {  
    while (fork() >= 0)  
        ;  
}
```

4. Explain what must happen for UNIX wait to return (successfully and) immediately.

3.2 Input/output

Computer systems have a wide diversity of input and output devices: keyboard, mouse, disk, USB port, Ethernet, WiFi, display, hardware timer, microphone, camera, accelerometer, and GPS, to name a few.

To deal with this diversity, we could specialize the application programming interface for each device, customizing it to the device’s specific characteristics.

After all, a disk device is quite different from a network and both are quite different from a keyboard: a disk is addressed in fixed sized chunks, while a network sends and receives a stream of variable sized packets, and the keyboard returns individual characters as keys are pressed. While the disk only returns data when asked, the network and keyboard provide data unprompted. Early computer systems took the approach of specializing the interface to the device, but it had a significant downside: every time a new type of hardware device is invented, the system call interface has to be upgraded to handle that device.

One of the primary innovations in UNIX was to regularize all device input and output behind a single common interface. In fact, UNIX took this one giant step further: it uses this same interface for reading and writing files and for interprocess communication. This approach was so successful that it is almost universally followed in systems today. We will sketch the interface in this section, and then in the next section, show how it can be used to build a shell.

The basic ideas in the UNIX I/O interface are:

- **Uniformity.** All device I/O, file operations, and interprocess communication use the same set of system calls: open, close, read and write.
- **Open before use.** Before an application does I/O, it must first call `open` on the device, file, or communication channel. This gives the operating system a chance to check access permissions and to set up any internal bookkeeping. Some devices, such as a printer, only allow one application access at a time — the `open` call can return an error if the device is in use. Open returns a handle to be used in later calls to read, write and close to identify the file, device or channel; this handle is somewhat misleadingly called a “file descriptor”, even when it refers to a device or channel so there is no file involved. For convenience, the UNIX shell starts applications with open file descriptors for reading and writing to the terminal.
- **Byte-oriented.** All devices, even those that transfer fixed-size blocks of data, are accessed with byte arrays. Similarly, file and communication channel access is in terms of bytes, even though we store data structures in files and send data structures across channels.
- **Kernel-buffered reads.** Stream data, such as from the network or keyboard, is stored in a kernel buffer and returned to the application on request. This allows the UNIX system call `read` interface to be the same for devices with streaming reads as those with block reads, such as disks and Flash memory. In both cases, if no data is available to be returned immediately, the `read` call blocks until it arrives, potentially giving up the processor to some other task with work to do.
- **Kernel-buffered writes.** Likewise, outgoing data is stored in a kernel buffer for transmission when the device becomes available. In the normal case, the system call `write` copies the data into the kernel buffer and

Open vs. creat vs. stat

By default, the UNIX `open` system call returns an error if the application tries to open a file that does not exist; as an option (not shown above), a parameter can tell the kernel to instead create the file if it does not exist. Since UNIX also has system calls for creating a file (`creat`) and for testing whether a file exists (`stat`), it might seem like `open` could be simplified to always assume that the file already exists.

However, UNIX often runs in a multi-user, multi-application environment, and in that setting the issue of system call design can become more subtle. Suppose instead of the UNIX interface, we had completely separate functions for testing if a file exists, creating a file, and opening the file. Assuming that the user has permission to test, open, or create the file, does this code work?

```
if (!exists(file)) {    // if the file doesn't exist
    create(file);      // create it; are we guaranteed the file doesn't exist?
}
open(file)           // open the file; are we guaranteed the file does exist?
```

The problem is that on a multi-user system, some other user might have created the file in between the call to test for its existence, and the call to create the file. Thus, call to create must also test the existence of the file. Likewise, some other user might have deleted the file between the call to create and the call to open. So `open` also needs the ability to test if the file is there, and if not to create the file (if that is the user's intent).

UNIX addresses this with an all-purpose, atomic `open`: test if the file exists, optionally create it if it doesn't, and then open it. Because system calls are implemented in the kernel, the operating system can make `open` (and all other I/O systems calls) non-interruptable with respect to other system calls. If another user tries to delete a file while the kernel is executing an `open` system call on the same file, the delete will be delayed until the `open` completes. The `open` will return a file descriptor that will continue to work until the application closes the file. The delete will remove the file from the file system, but the file system does not actually reclaim its disk blocks until the file is closed.

returns immediately. This decouples the application from the device, allowing each to go at their own speed. If the application generates data faster than the device can receive it (as is common when spooling data to a printer), the `write` system call blocks in the kernel until there is enough room to store the new data in the buffer.

- **Explicit close.** When an application is done with the device or file, it calls `close`. This signals to the operating system that it can decrement the reference-count on the device, and garbage collect any unused kernel data structures.

For interprocess communication, we need two more system calls:

- **Pipes.** A UNIX pipe is a kernel buffer with two file descriptors, one for writing (to put data into the pipe) and one for reading (to pull data out

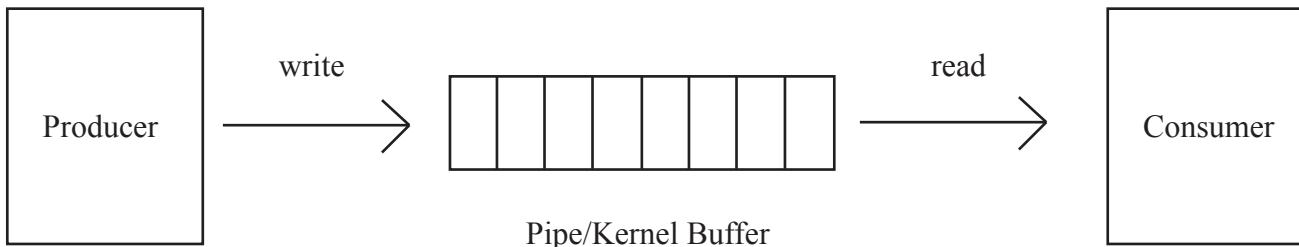


Figure 3.6: A pipe is a temporary kernel data structure to connect a process producing data with a process consuming the data.

of the pipe), as illustrated in Figure 3.6. Data is read in exactly the same sequence it is written, but since the data is buffered, the execution of the producer and consumer can be decoupled, reducing waiting in the common case. The pipe terminates when either endpoint closes the pipe or exits.

The Internet has a similar facility to UNIX pipes called TCP (Transmission Control Protocol). Where UNIX pipes connect processes on the same machine, TCP provides a bi-directional pipe between two processes running on different machines. In TCP, data is written as a sequence of bytes on one machine, and read out as the same sequence on the other machine. We plan to discuss the implementation of TCP in detail in a chapter to be added to the next revision of this textbook.

- **Replace file descriptor.** By manipulating the file descriptors of the child process, the shell can cause the child to read its input from, or send its output to, a file or a pipe instead of from a keyboard or to the screen. This way, the child process does not need to be aware of who is providing or consuming its I/O. The shell does this redirection using a special system call named `dup2(from, to)` that replaces the `to` file descriptor with a copy of the `from` file descriptor.
- **Wait for multiple reads.** For client-server computing, a server may have a pipe open to multiple client processes. Normally, read will block if there is no data to be read, and it would be inefficient for the server to poll each pipe in turn to check if there is work for it to do. The UNIX system call `select(fd[], number)` addresses this. Select allows the server to wait for input from any of a set of file descriptors; it returns the file descriptor that has data, but it does not read the data. Windows has an equivalent function, called `WaitForMultipleObjects`.

Figure 3.7 summarizes the dozen UNIX system calls discussed in this section.

```
\begin{itemize}
// Create a child process as a clone of the current process;
// fork returns to both the parent and child
fork()

// Run the application "prog" in the current process
exec(prog, args)

// Tell the kernel the current process is complete,
// and its data structures should be garbage collected
exit()

// Pause until the child process has exited
wait(process_ID)

// Send an interrupt of "type" to a process
signal(process_ID, type)

// Open a file or hardware device, specified by "name";
// returns a file descriptor that can be used by other calls
fd = open(name)

// Create a one-directional pipe between two processes;
// returns two file descriptors, one for reading, one for writing
pipe(fd[2])

// Replace the to_fd file descriptor with a copy of from_fd;
// used for replacing stdin/stdout
dup2(from_fd, to_fd)

// Read up to "size" bytes into buffer, from the device, file or channel.
// "read" returns the number of bytes actually read; for streaming
// devices this will often be less than "size".
// For example, a read from the keyboard device will return all of its queued bytes.
int read(fd, buffer, size)

// Analogous to "read", write up to "size" bytes into kernel output
// buffer for a device, file or channel.
// "write" normally returns immediately, but may stall if there is
// no space in the kernel buffer.
int write(fd, buffer, size)

// Return when any of the file descriptors in the array have data available to be read.
// Returns the file descriptor with the data.
fd = select(fd[], number)

// Tell the kernel the process is done with this device, file, or channel.
close(fd)
\end{itemize}
```

Figure 3.7: List of UNIX system calls discussed in this section.

```

main() {
    char *prog = NULL;
    char **args = NULL;

    // read the next line from the input, and parse it into the program name and its arguments
    // return false if we've reached the end of the input
    while(readAndParseCmdLine(&prog, &args)) {

        int child_pid = fork();           // create a child process to run the command

        if (child_pid == 0) {            // I'm the child process
            exec(prog, args);          // child uses the parent's input and output
            // NOT REACHED
        } else {                        // I'm the parent
            wait(child_pid);           // wait for the child to complete
            return 0;
        }
    }
}

```

Figure 3.8: Example code for a simple UNIX shell.

3.3 Case Study: Implementing a shell

The dozen UNIX system calls listed above are enough to build a flexible and powerful command line shell, one that runs entirely at user-level with no special permissions. As we mentioned, the process that creates the shell is responsible for providing it an open file descriptor for reading commands for its input (e.g., from the keyboard), called *stdin* and for writing output (e.g., to the display), called *stdout*.

Definition: **stdin**
Definition: **stdout**

Figure 3.8 illustrates the code for the basic operation of a shell. The shell reads a command line from the input, and it forks a process to execute that command. UNIX `fork` automatically duplicates all open file descriptors in the parent, incrementing the kernel’s reference counts for those descriptors, so the input and output of the child is the same as the parent. The parent waits for the child to finish before it reads the next command to execute.

Because the commands to read and write to an open file descriptor are the same whether the file descriptor represents a keyboard, screen, file, device, or pipe, UNIX programs do not need to be aware of where their input is coming from, or where their output is going. This is helpful in a number of ways:

- **A program can be a file of commands.** Programs are normally a set of machine instructions, but on UNIX a program can be a file containing a list of commands for a shell to interpret. To disambiguate, shell programs signified in UNIX by putting “#! interpreter” as the first line of the file, where “interpreter” is the name of the shell executable.

The UNIX C compiler is structured this way. When it is exec’ed, the kernel recognizes it as a shell file and starts the interpreter, passing it the

file as input. The shell reads the file as a list of commands to invoke the pre-processor, parser, code generator and assembler in turn, exactly as if it was reading text input from the keyboard. When the last command completes, the interpreter exits to inform the kernel that the program is done.

- **A program can send its output to a file.** By changing the stdout file descriptor in the child, the shell can redirect the child's output to a file. In the standard UNIX shell, this is signified with a “greater than” symbol. Thus, “ls > tmp” lists the contents of the current directory into the file “tmp”. After the fork and before the exec, the shell can replace the stdout file descriptor for the child using dup2. Because the parent has been cloned, changing stdout for the child has no effect on the parent.
- **A program can read its output from a file.** Likewise, by using dup2 to change the stdin file descriptor, the shell can cause the child to read its input from a file. In the standard UNIX shell, this is signified with a “less than” symbol. Thus, “zork < solution” plays the game “zork” with a list of instructions stored in the file “solution.”
- **The output of one program can be the input to another program.** The shell can use a pipe to connect two programs together, so that the output of one is the input of another. This is called a *producer-consumer* relationship. For example, in the C-compiler, the output of the pre-processor is sent to the parser, and the output of the parser is sent to the code-generator and then to the assembler. In the standard UNIX shell, a pipe connecting two programs is signified by a “—” symbol, as in: “cpp file.c — cparse — cgen — as & file.o”. In this case the shell creates four separate child processes, each connected by pipes to its predecessor and successor. Each of the phases can run in parallel, with the parent waiting for all of them to finish.

Definition:
producer-consumer

Exercises

5. Suppose you were the instructor of a very large introductory programming class. Explain (in English) how you would use UNIX system calls to automate testing of submitted homework assignments.
6. What happens if you run “exec csh” in a UNIX shell? Why?
7. What happens if you run “exec ls” in a UNIX shell? Why?

3.4 Case Study: Interprocess communication

For many of the same reasons it makes sense to construct complex applications from simpler modules, it often makes sense to create applications that can specialize on a specific task, and then combine those applications into more complex structures. We gave an example above with the C compiler, but many parts of the operating system are structured this way. For example, instead of every program needing to know how to coordinate access to a printer, UNIX has a printer server, a specialized program for managing the printer queue.

For this to work, we need a way for processes to communicate with each other. Three widely used forms of interprocess communication are:

- **Producer-consumer.** In this model, programs are structured to accept as input the output of other programs. Communication is one-way: the producer only writes, and the consumer only reads. As we explained above, this allows chaining: a consumer can be in turn a producer for a different process. Much of the success of UNIX was due to its ability to easily compose many different programs together in this fashion.
- **Client-server.** An alternative model is to allow two-way communication between processes, as in client-server computing. The server implements some specialized task, such as managing the printer queue or managing the display. Clients send requests to the server to do some task, and when the operation is complete, the server replies back to the client.
- **File system.** Another way programs can be connected together is through reading and writing files. A text editor can import an image created by a drawing program, and the editor can in turn write an HTML file that a web server can read to know how to display a web page. A key distinction is that, unlike the first two modes, communication through the file system can be separated in *time*: the writer of the file does not need to be running at the same time as the file reader. Therefore data needs to be stored persistently on disk or other stable storage, and the data needs to be named so that you can find the file when needed later on.

All three models are widely used both on a single system and over a network. For example, the Google mapreduce utility is a central part of many of Google's services, and it operates over a network in a producer-consumer fashion: the output of the map function is sent to the machines running the reduce function. The web is an example of client-server computing, and many enterprises and universities run centralized file servers to connect a text editor on one computer with a compiler running on another.

As persistent storage, file naming, and distributed computing are each complex topics in their own right, we defer the discussions of those topics to later chapters. Here we focus on interprocess communication, where both processes are running simultaneously on the same machine.

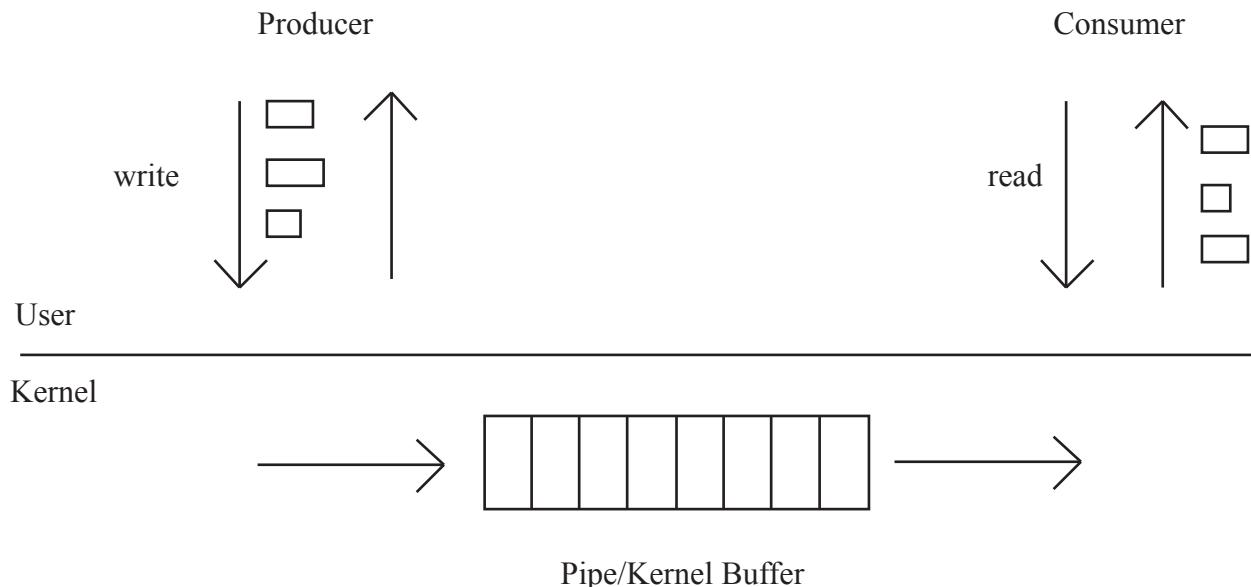


Figure 3.9: Interprocess communication between a producer application and a consumer.

3.4.1 Producer-consumer communication

Figure 3.9 illustrates how two processes communicate through the operating system in a producer-consumer relationship. Via the shell, we establish a pipe between the producer and the consumer. As one process computes and produces a stream of output data, it issues a sequence of write system calls on the pipe into the kernel. Each write can be of variable size. Assuming there is room in the kernel buffer, the kernel copies the data into the buffer, and returns immediately back to the producer.

At some point later, the operating system will schedule the consumer process to run. (On a multicore, the producer and consumer could be running at the same time.) The consumer issues a sequence of read calls. Because the pipe is just a stream of bytes, the consumer can read the data out in any convenient chunking — the consumer can read chunks in 1KB chunks, while the producer wrote its data in 4KB chunks, or vice versa. Each system call read made by the consumer returns the next successive chunk of data out of the kernel buffer. The consumer process can then compute on its input, sending its output to the display, a file, or onto the next consumer.

The kernel buffer allows each process to run at its own pace. There is no requirement that each process have equivalent amounts of work to do. If the producer is faster than the consumer, the kernel buffer fills up, and when the producer tries to write to a full buffer, the kernel stalls the process until there

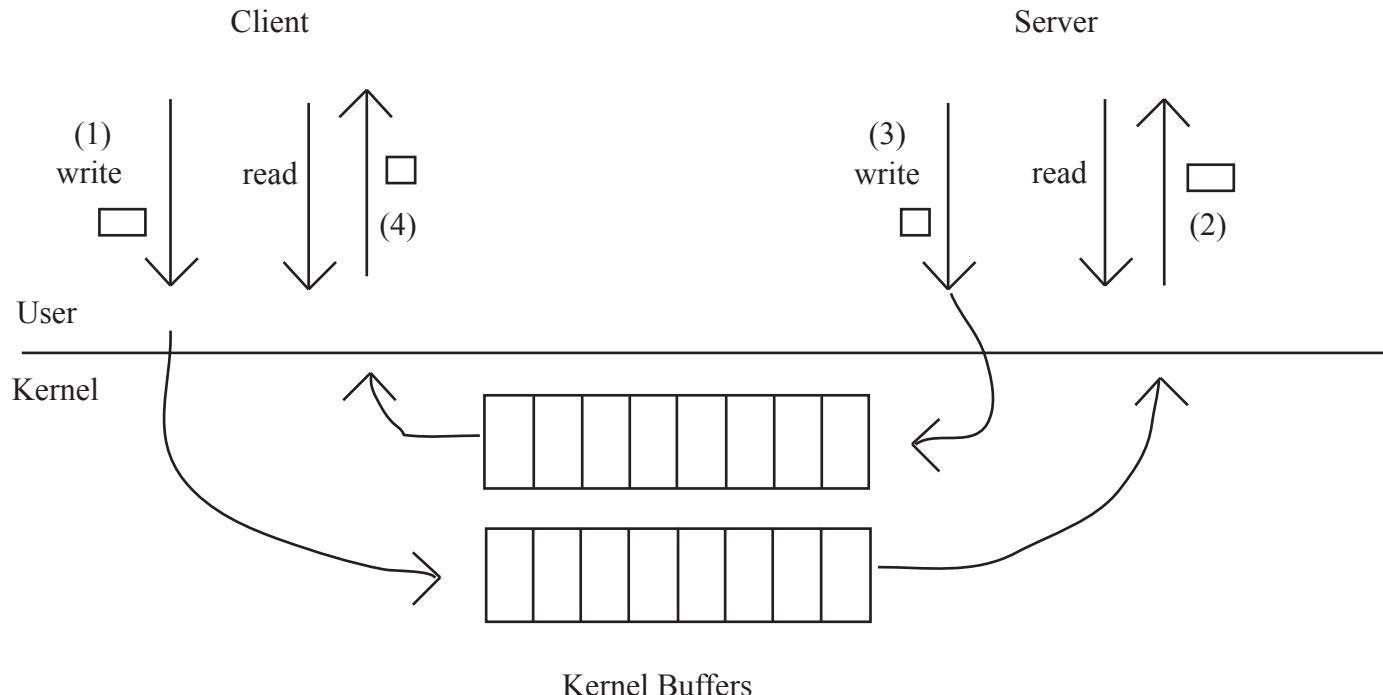


Figure 3.10: Interprocess communication between a client process and a server process.

is room to store the data. Equivalently, if the consumer is faster than the producer, the buffer will empty and the next read request will stall until the producer creates more data.

In UNIX, when the producer finishes, it closes its side of the pipe, but there may still be data queued in the kernel for the consumer. Eventually, the consumer reads the last of the data, and the read system call will return an “end of file” marker. Thus, to the consumer, there is no difference between reading from a pipe and reading from a file.

Decoupling the execution of the producer and consumer through the use of kernel buffers reduces the number and cost of context switches. Modern computers make extensive use of hardware caches to improve performance, but caches are ineffective if a program only runs for a short period of time before it must yield the processor to another task. The kernel buffer allows the operating system to run each process long enough to benefit from reuse, rather than alternating between the producer and consumer on each system call.

```

Client:
    char request[RequestSize];
    char reply[ReplySize]

    // ...compute..
    // put the request into the buffer
    // send the buffer to the server
    write(output, request, RequestSize);

    // wait for response
    read(input, reply, ReplySize);

    // ...compute..

Server:
    char request[RequestSize];
    char reply[ReplySize];

    // loop waiting for requests
    while (1) {
        // read incoming command
        read(input, request, RequestSize);

        // do operation

        // send result
        write(output, reply, ReplySize);
    }

```

Figure 3.11: Example code for client-server interaction.

3.4.2 Client-server communication

We can generalize the above to illustrate client-server communication, shown in Figure 3.10. Instead of a single pipe, we create two, one for each direction. To make a request, the client writes the data into one pipe, and reads the response from the other. The server does the opposite: it reads requests from the first pipe, performs whatever is requested (provided the client has permission to make the request), and writes the response onto the second pipe.

The client and server code are shown in Figure 3.11. To simplify the code, we assume that the requests and responses are fixed size.

Frequently, we want to allow many clients to talk to the same server. For example, there is one server to manage the print queue, although there can be many processes that want to be able to print. For this, the server uses the `select` system call, to identify the pipe containing the request to be read, as shown in Figure 3.12. The client code is unchanged.

3.5 Operating system structure

We started this chapter with a list of functionality that users and applications need from the operating system. We have shown that by careful design of the

Streamlining client-server communication

Client-server communication is a common pattern in many systems, and so one can ask: how can we improve its performance? One step is to recognize that both the client and the server issue a `write` immediately followed by a `read`, to wait for the other side to reply; at the cost of adding a system call, these can be combined to eliminate two kernel crossings per round trip. Further, the client will always need to wait for the server, so it makes sense for it to donate its processor to run the server code, reducing delay. Microsoft added support for this optimization to Windows in the early 1990's when it converted to a microkernel design (explained a bit later in this chapter). However, as we noted earlier, modern computer architectures make extensive use of caches, so for this to work we need code and data for both the client and the server to be able to be in cache simultaneously. We will talk about mechanisms to accomplish that in a later chapter.

We can take this streamlining even further. On a multicore system, it is possible or even likely that both the client and server each have their own processor. If the kernel sets up a shared memory region accessible to both the client and the server and no other processes, then the client and server can (safely) pass requests and replies back and forth, as fast as the memory system will allow, without ever traversing into the kernel or relinquishing their processors.

```
Server:
char request[RequestSize];
char reply[ReplySize];
FileDescriptor clientInput[NumClients];
FileDescriptor clientOutput[NumClients];

// loop waiting for a request from any client
while (fd = select(clientInput, NumClients)) {
    // read incoming command from a specific client
    read(clientInput[fd], request, RequestSize);

    // do operation

    // send result
    write(clientOutput[fd], reply, ReplySize);
}
```

Figure 3.12: Server code for communicating with multiple clients.

system call interface, we can offload some of the work of the operating system to user programs, such as to a shell or to a print server.

In the rest of this chapter, we ask how should we organize the remaining parts of the operating system? There are many dependencies among the modules inside the operating system, and there is often quite frequent interaction between these modules:

- Many parts of the operating system depend on synchronization primitives for coordinating access to shared data structures with the kernel.
- The virtual memory system depends on low level hardware support for address translation, support that is specific to a particular processor architecture.
- Both the file system and the virtual memory system share a common pool of blocks of physical memory. They also both depend on the disk device driver.
- The file system can depend on the network protocol stack if the disk is physically located on a different machine.

This has led operating system designers to wrestle with a fundamental trade-off: by centralizing functionality in the kernel, performance is improved and it makes it easier to arrange tight integration between kernel modules. However, the resulting systems are less flexible, less easy to change, and less adaptive to user or application needs. We discuss these tradeoffs by describing several options for the operating system architecture.

3.5.1 Monolithic kernels

Definition: **monolithic kernel**

Almost all widely used commercial operating systems, such as Windows, macOS, and Linux, take a similar approach to the architecture of the kernel — a monolithic design. As shown in Figure 3.13, with a *monolithic kernel*, most of the operating system functionality runs inside the operating system kernel. In truth, the term is a bit of a misnomer, because even in so-called monolithic systems, there are often large segments of what users consider the operating system that runs outside the kernel, either as utilities like the shell, or in system libraries, such as libraries to manage the user interface.

Internal to a monolithic kernel, the operating system designer is free to develop whatever interfaces between modules that make sense, and so there is quite a bit of variation from operating system to operating system in those internal structures. However, two common themes emerge across systems: to improve portability, almost all modern operating systems have both a hardware abstraction layer and dynamically loaded device drivers.

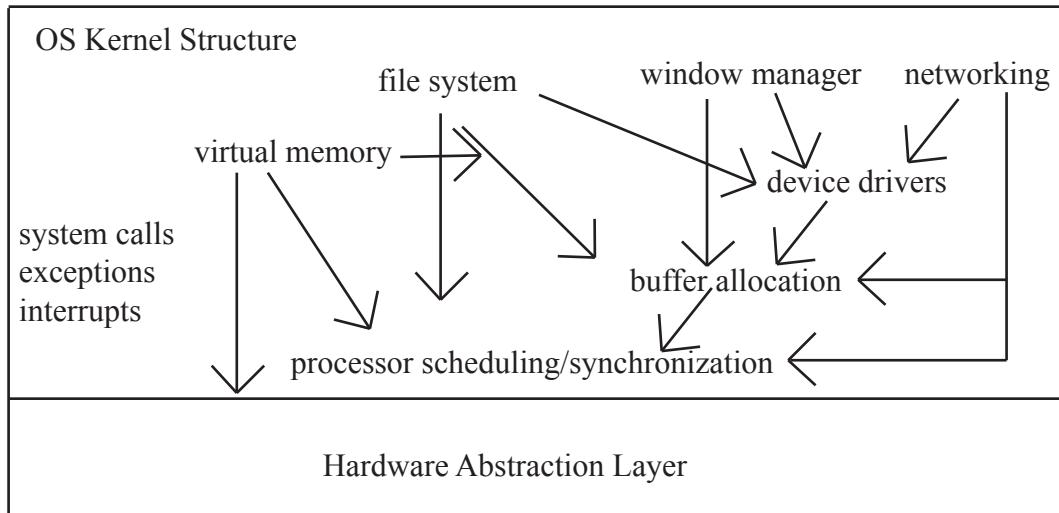


Figure 3.13: Interprocess communication between a client process and a server process.

Hardware abstraction layer

A key goal of operating systems is to be portable across a wide variety of hardware platforms. To accomplish this, especially within a monolithic system, requires careful design of the *hardware abstraction layer*. The hardware abstraction layer (HAL) is a portable interface to machine-specific operations within the kernel. For example, almost all general-purpose operating systems perform process and thread context switches, but the specific implementation of those routines will vary depending on the processor architecture. The exception, interrupt, and system call trap handling is also machine-specific; all systems have those functions, but the specific implementation will vary. As we will see in a later chapter, machines differ quite a bit in their architecture for managing virtual address spaces; most kernels provide portable abstractions on top of the machine-dependent routines, such as to translate virtual addresses to physical addresses or to copy memory from applications to kernel memory and vice versa.

Definition: **hardware abstraction layer**

With a well-defined hardware abstraction layer in place, most of the operating system is machine-independent. Thus, porting an operating system to a new processor architecture is just a matter of creating new implementations of these low level HAL routines, and recompiling.

Dynamically installed device drivers

A similar consideration leads to operating systems that can easily accommodate a wide variety of physical I/O devices. Although there are only a handful of

different instruction set architectures in wide use today, there are a huge number of different types of physical I/O devices, manufactured by a large number of companies. There is diversity in the hardware interfaces to devices as well as in the hardware chip sets for managing the devices. A recent survey found that approximately 70% of the code in the Linux kernel was in device-specific software.

To keep the rest of the operating system kernel portable, we want to decouple the operating system source code from the specifics of each device. For instance, suppose a manufacturer creates a new model a printer — what steps are needed by the operating system manufacturer to accommodate that change?

Definition: **dynamically loadable device driver**

The key innovation, widely adopted today, is a *dynamically loadable device driver*. A dynamically loadable device driver is software to manage a specific device or interface or chipset, that is added to the operating system kernel after the kernel starts running, to handle the devices that are present on a particular machine. The device driver code is typically written by the device manufacturer, using a standard interface provided by the kernel. The operating system kernel calls into the driver whenever it needs to read or write data to the device.

At boot, the operating system starts with a small number of device drivers – e.g., for the disk (to read the operating system binary into memory). For the devices physically attached to the computer, the computer manufacturer bundles those drivers into a file it stores along with the bootloader. When the operating system starts up, it queries the I/O bus for which devices are attached to the computer and then loads those drivers from the file on disk. Finally, for any network-attached devices, such as a network printer, the operating system can load those drivers over the Internet.

While dynamically loadable device drivers solve one problem, they pose a different one. Errors in a device driver can corrupt the operating system kernel and application data structures; just as with a regular program, errors may not be caught immediately, so that user may be unaware that their data is being silently modified. Even worse, a malicious attacker can use device drivers to introduce a computer virus into the operating system kernel, and thereby silently gain control over the entire computer. Recent studies have found that 90% of all system crashes were due to bugs in device drivers, rather than in the operating system itself.

Operating system developers have taken three approaches to dealing with this issue:

- **Code inspection.** Operating system vendors typically require all device driver code to be submitted in advance for inspection and testing, before being allowed into the kernel.
- **Bug tracking.** After every system crash, the operating system can collect information about the system configuration and the current kernel stack, and sends this information back to a central database for analysis.

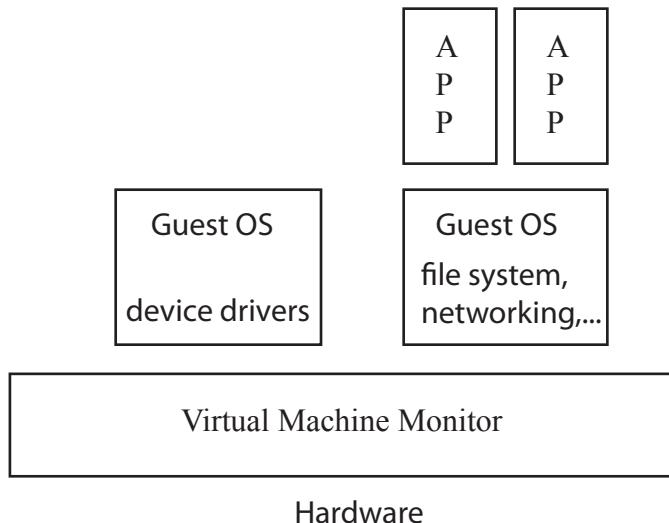


Figure 3.14: Legacy device drivers can be run inside a guest operating system on top of a virtual machine in order to isolate the effect of implementation errors in driver code.

Microsoft does this on a wide scale. With hundreds of millions of installed computers, even a low rate of failure can yield millions of bug reports per day. Many crashes happen inside the device driver itself, but even those that do not can sometimes be tracked down. For example, if failures are correlated with the presence of a particular device driver, or increase after the release of a new version of the driver, that can indicate the source of a problem.

- **Driver sandboxing.** Some researchers have proposed running device drivers in their own restricted execution environment, using some of the techniques described in the previous chapter for implementing application process isolation. This way, a buggy driver can only affect its own internal data structures and not the rest of the operating system kernel. Of course, this requires lightweight sandboxing techniques, a topic we'll return to in a later chapter.

Although driver sandboxing is likely to be adopted over the long term, in the short term it has proven difficult to implement in current operating systems. There is a huge amount of existing device driver code that makes use of the ability to directly address internal kernel data structures; drawing a boundary around these drivers has proven difficult. Supporting legacy drivers is likely to be less of a problem as completely new hardware and operating system platforms, such as smartphones and tablets, are developed.

In the meantime, one approach that has gained some traction is to run all

device driver code inside a guest operating system running on a virtual machine, as shown in Figure 3.14. The guest operating system loads the device drivers as if it was running directly on the real hardware, but when the devices attempt to access the physical hardware, the underlying virtual machine monitor regains control to ensure safety. Device drivers can still have bugs, but they can only corrupt the guest operating system and not other applications running on the underlying virtual machine monitor.

3.5.2 Microkernel

An alternative to the monolithic kernel approach is to run as much of the operating system as possible in one or more user-level servers. The window manager on most operating systems is implemented this way: individual applications draw items on their portion of the screen by sending requests to the window manager. The window manager adjudicates which application window is in front or in back for each pixel on the screen, and then renders the result. If the system has a hardware graphics accelerator present, the window manager can use it to render items more quickly. Some systems have moved other parts of the operating system into user-level servers: the network stack, the file system, device drivers, and so forth.

The difference between a monolithic and a microkernel design is often transparent to the application programmer. The location of the service can be hidden in a user-level library — calls go to the library, which casts the requests either as system calls or as reads and writes to the server through a pipe. The location of the server can also be hidden inside the kernel — the application calls the kernel as if the kernel implements the service, but instead the kernel reformats the request into a pipe that the server can read.

A microkernel design offers considerable benefit to the operating system developer, as it easier to modularize and debug user-level services than kernel code. Aside from a potential reliability improvement, however, microkernels offer little in the way of visible benefit to end users and can slow down overall performance by inserting extra steps between the application and the services it needs. Thus, in practice, most systems adopt a hybrid model where some operating system services are run at user-level and some are in the kernel, depending on the specific tradeoff between code complexity and performance.

3.6 Conclusion and future directions

In this chapter, we have seen how system calls can be used by applications to create and manage processes, perform I/O, and communicate with other processes. Every operating system has its own unique system call interface; describing even a single interface in depth would be beyond the scope of this

book. In this chapter, we focused parts of the UNIX interface because it is both compact and powerful. A key aspect of the UNIX interface are that creating a process (with `fork`) is separate from starting to run a program in that process (with `exec`); another key feature is the use of kernel buffers to decouple reading and writing data through the kernel.

Operating systems use the system call interface to provide services to applications and to aid in the internal structuring of the operating system itself. Almost all general-purpose computer systems today have a user-level shell and/or a window manager that can start and manage applications on behalf of the user. Many systems also implement parts of the operating system as user-level services accessed through kernel pipes.

As we've noted, a trend is for applications to become mini-operating systems in their own right, with multiple users, resource sharing and allocation, untrusted third-party code, processor and memory management, and so forth. The system call interfaces for Windows and UNIX were not designed with this in mind, and an interesting question is how they will change to accommodate this future of powerful meta-applications.

In addition to the fine-grained sandboxing and process creation we described at the end of the last chapter, a trend is to re-structure the system call interface to make resource allocation decisions explicit and visible to applications. Traditionally, operating systems make resource allocation decisions — when to schedule a process or a thread, how much memory to give a particular application, where and when to store its disk blocks, when to send its network packets — transparently to the application, with a goal of improving end user and overall system performance. Applications are unaware of how many resources they have, appearing to run by themselves, isolated on their own (virtual) machine.

Of course, the reality is often quite different. An alternate model is for operating systems to divide resources among applications and then allow each application to decide for itself how best to use those resources. One can think of this as a type of federalism. If both the operating system and applications are governments doing their own resource allocation, they are likely to get in each other's way if they aren't careful. As a simple example, consider how a garbage collector works; it assumes it has a fixed amount of memory to manage. However, as other applications start or stop, it can gain or lose memory, and if the operating system does this reallocation transparently, the garbage collector has no hope of adapting. Later in the book, we will see examples of this same design pattern in many different areas of operating system design.

Exercises

For convenience, the exercises from the body of the chapter are repeated here.

1. Can UNIX `fork` return an error? Why or why not?

Note: You can answer this question by looking at the manual page for fork, but before you do that, think about what the fork system call does. If you were designing this call, would you need to allow fork to return an error?

Note: A manual page (or “man page”) is a standard way of documenting Unix system calls and utility programs. On a Unix machine, you can access a manual page by running the `man` command (e.g., `man fork`). Another way to find manual pages is via web search (e.g., search for `man fork` and many of the top results will be manual pages for the fork system call.)

2. Can UNIX exec return an error? Why or why not?

Note: You can answer this question by looking at the manual page for exec, but before you do that, think about what the exec system call does. If you were designing this call, would you need to allow it to return an error?

3. What happens if we run the following program on UNIX?

```
main() {
    while (fork() >= 0)
        ;
}
```

4. Explain what must happen for UNIX wait to return (successfully and) immediately.

5. Suppose you were the instructor of a very large introductory programming class. Explain (in English) how you would use UNIX system calls to automate testing of submitted homework assignments.

6. What happens if you run “exec csh” in a UNIX shell? Why?

7. What happens if you run “exec ls” in a UNIX shell? Why?

8. Consider the following program:

```
main(int argc, char ** argv) {
    forkthem(5)
}

void forkthem(int n) {
    if(n > 0) {
        fork();
        forkthem(n-1);
    }
}
```

How many processes are created if the above piece of code is run?

9. Consider the following program:

```
main(int argc, char ** argv) {
    int child = fork();
    int x = 5;

    if (child == 0) {
        x += 5;
    } else {
        child = fork();
        x += 10;
        if(child) {
            x += 5;
        }
    }
}
```

How many different copies of the variable `x` are there? What are their values when their process finishes?

10. What is the output of the following programs? (Please try to solve the problem without compiling and running the program.)

```
// Program 1
main() {
    int val = 5;

    if(fork())
        wait(&val);
    val++;
    printf("%d\n", val);
    return val;
}

// Program 2:
main() {
    int val = 5;

    if(fork())
        wait(&val);
    else
        exit(val);
    val++;
    printf("%d\n", val);
    return val;
}
```

11. Implement a shell using UNIX system calls. Features of your shell can include pipes, background and foreground tasks, and job control (suspend, resume, and kill).

Part II

Concurrency

Chapter 4

Concurrency and Threads

Many hands make light work.

— anon

In the real world—outside of computers—different activities often proceed at the same time: five jazz musicians play their instruments while reacting to each other; one car drives north while another drives south; one part of a drug molecule is attracted to a cell’s receptor while another part is repelled; a humanoid robot walks, raises its arms, and turns its head; I fetch one article from the *The New York Times* website while you fetch another; or millions of people make long distance phone calls on Mother’s Day.

Internally, computers also harness concurrency. For example, a modern server might have 8 processors, 10 disks, and 4 network interfaces; a workstation might have a dozen active IO devices including a screen, a keyboard, a mouse, a camera, a microphone, a speaker, a wireless network interface, a wired network interface, an internal disk drive, an external disk drive, a printer, and a scanner; today, even mobile phones can have dual-core processors.

However, programmers are used to thinking sequentially. For example, when reading or writing the code for a procedure, we identify an initial state and a set of preconditions, think through how each successive statement changes the state, and from that determine the postconditions. How would one even think about a program where dozens of things are happening at once?

To simulate or interact with the real world, to manage hardware resources, and to map parallel applications to parallel hardware, computer systems must provide programmers with abstractions for expressing and managing *concurrency*. These abstractions are used both by applications and within the operating system, itself.

Threads: A core abstraction for concurrency. This chapter will focus on the powerful concurrency abstraction of *threads*. Threads let us define a set of tasks that should run at the same time, but they let us write the code for each task as if the task were standard, sequential code.

Threads are used within the operating system and within user-level processes.

Example: Threads in an operating system. For example an operating system kernel may have one thread that writes modified file blocks from memory to disk, another thread that encrypts (or decrypts) blocks as they are written to (or read from) an encrypted disk, another thread that monitors system load and manages the processor's power-saving features, and another thread that scans the cache of disk blocks and frees old entries.

Example: Threads in an application. Consider an Earth Visualizer application similar to Google Earth (<http://earth.google.com/>). This application allows a user to virtually fly anywhere in the world, to see aerial images at different resolutions, and to view other information associated with each location. A key part of the design is that the user's controls are always operable so that as the user mouses to a new location, the image is redrawn in the background at successively better resolutions while the program continues to allow the user to adjust the view, to select additional information about the location for display, or to enter search terms.

To implement this application, as Figure 4.1 illustrates the programmer might write code to draw a portion of the screen, code to draw user interface (UI) widgets and process user inputs, and code to fetch higher resolution images for newly-visible areas. In a sequential program, these functions would run in turn. With threads, these can run concurrently so that the user interface is responsive even while new data is being fetched and the screen is being redrawn

Chapter overview. The rest of this chapter discusses three topics in detail:

- **Thread abstraction.** What are threads and what do they do?
- **Thread internals.** What building blocks are needed so that we can construct threads?
- **Thread implementation.** Given these building blocks, how can we implement threads?

4.1 Threads: Abstraction and interface

As Figure 4.2 illustrates, threads *virtualize the processor*, providing the illusion that programs run on machines with an infinite number of virtual processors.



Figure 4.1: In the Earth Visualizer example, two threads each draw part of the scene, a third thread manages the user interface widgets, and a fourth thread fetches new data from a remote server. Satellite Image Credit: NASA Earth Observatory.

The programs can then create however many threads they need without worrying about the exact number of physical processors, and each thread runs on its own virtual processor.

Of course, the physical reality is that a given system only has so many processors. The operating system's job is to multiplex all of the system's threads on the actual physical processor present in the system. It does this by transparently suspending and resuming threads so that at any given time only a subset of the threads are actively running.

Multi-threaded programs. The intuition for using the thread abstraction is simple: in a program, we can represent each concurrent task as a *thread*. Each thread provides the abstraction of a sequential execution similar to the traditional programming model. In fact, we can think of a traditional program as *single-threaded* with one logical sequence of steps as each instruction follows the previous one. The program executes statements, iterates through loops, and

Deja vu all over again?

Threads are widely used and modern programming languages often make it easy to use threads. You may have programmed with threads before or may have taken classes that talk about using threads. What is new here?

The discussions in this chapter and the next are designed to make sense even if you have never seen threads before. If you have seen threads before, great! But we think you will find the discussions useful.

Beyond describing the basics threads abstractions, we emphasize two points in our discussions in this chapter and the next the next one.

- **Implementation.** We will describe how operating systems implement threads both for their own use and for use by user-level applications. It is important to understand how threads really work so that you can understand their costs and performance characteristics and use them effectively.
- **Practice.** We will present a methodology for writing correct multi-threaded programs. Concurrency is increasingly important in almost all significant programming projects, but writing correct multi-threaded programs requires much more care and discipline than writing correct single-threaded programs. That said, following a few simple rules that we will describe can greatly simplify the process of writing robust multi-threaded code.

Multithreaded programming has a reputation for being difficult, and we believe the ideas in this chapter and the next can help almost anyone become better at programming with threads.

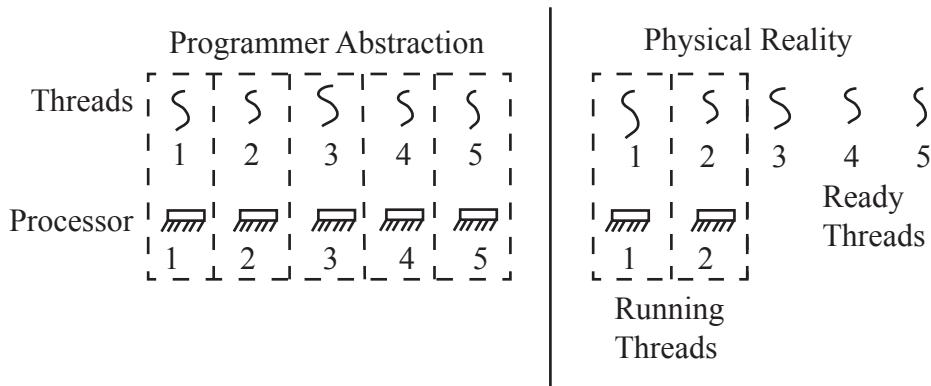


Figure 4.2: Threads virtualize the processor, providing the illusion that the system has an infinite number of processors. Then, programmers can create as many threads as they need, and each thread runs on a virtual processor. In reality, of course, a machine only has a finite number of processors, and it is the operating system's job to transparently multiplex threads on the actual processors.

calls/returns from procedures one after another.

A *multi-threaded program* is a generalization of the same basic programming model. Each individual thread follows a single sequence of steps as it executes statements, iterates through loops, calls/returns from procedures, etc. However, a program can now have several such threads executing at the same time.

Uses for threads. A programmer—whether programming the operating system or an outside application—uses threads to address three issues:

- **Program structure: Expressing logically concurrent tasks.** Programs often interact with or simulate real world applications that have concurrent activities. Threads allow the programmer to express an application’s concurrency by writing each concurrent task as a separate thread.

Example: Threads for program structure. In our Earth Visualizer application, threads allow different activities—updating the screen, fetching additional data, and receiving new user inputs—to run in at the same time. For example, we do not want to freeze the user interface widgets while we fetch data and redraw the screen.

Programmers are used to thinking sequentially, and threads allow them to write the code for each task as a separate, sequential program rather than requiring them to write one program that somehow interleaves all of these activities. Although one could imagine manually writing a program that interleaves these activities (e.g., draw a few pixels on the screen, then check to see if the user has moved the mouse, then check to see if new image data have arrived on the network, …), using threads can make things much simpler.

- **Performance: Exploiting multiple processors.** If programs can use multiple processors to do their processing in parallel, they can run faster—doing the same work in less time or doing more work in the same time. Today, a server may have over a dozen processors, a desktop or laptop machines may include two to eight processor cores, and even some cell phones are multicore machines. Looking forward, Moore’s law makes it likely that the number of processors per system will continue to increase.

An important property of the threads abstraction is that the number of threads used by the program need not match the number of processors in the hardware on which it is running. The operating system transparently switches which threads are running on which processors.

Example: Threads for parallel processing. If our Earth Visualizer application runs on an 8-processor machine, the programmer might parallelize the demanding job of rendering different portions of the image on the screen across seven threads. Then, the operating system might run those seven rendering threads on seven processors and run the remaining threads on the remaining processor to update the on-screen

navigation widgets, to construct the network messages needed to fetch additional images from the distant servers, and to parse the reply messages that come back.

- **Performance: Coping with high-latency I/O devices.** To do useful work, computers must interact with the outside world via Input/Output (I/O) devices. By running tasks as separate threads, when one task is waiting for I/O, the processor can make progress on a different task.

We do not want the processor to be idle while waiting for I/O to complete for two reasons.

First, processors are often much faster than the I/O systems with which they interact, so it is useful for a programmer to multiplex the processor among multiple tasks. For example, the latency to read from disk can be tens of milliseconds, so after requesting a block from disk, an operating system switches to doing other work until the disk has copied that block to the machine's main memory.

Second, I/O provides a way for the computer to interact with external entities like users pressing keys on a keyboard or a remote computer sending network packets. The arrival of this type of I/O event is unpredictable, so we want the processor to be able to work on other things while still responding quickly to these events.

Example: Threads for I/O performance. In our Earth Visualizer application, a snappy user interface is essential, but much of the imagery is stored on remote servers and fetched by the application only when needed. The application still provides a responsive experience when a user changes location by simultaneously drawing a low-resolution view of the new location while fetching higher-resolution images from the distant servers. The application then progressively updates the view as the higher-resolution images arrive.

4.1.1 Threads: Definition and properties

Above, we sketched what a thread is, how it is used, and why it is useful. To go further, we must define a thread and its properties more precisely.

Definition: **thread** A *thread* is a *single execution sequence* that represents a *separately schedulable task*.

- **Single execution sequence.** By *single execution sequence* we mean that each thread executes a sequence of instructions—assignments, conditionals, loops, procedures, etc.—just as in the familiar programming model.
- **Separately schedulable task.** By *separately schedulable task* we mean that the operating system can run, suspend, or resume a thread at any time.

Running, suspending, and resuming threads. Threads provide the illusion of an infinite number of processors. How does the operating system implement this illusion? It needs to be able to execute instructions from each thread so that each thread makes progress, but the underlying hardware has only a limited number of processors—maybe only one!

To map an arbitrary set of threads to a fixed set of processors, operating systems include a *scheduler* that can switch which threads are running and which are ready but not running. For example, in Figure 4.2, a scheduler might suspend thread 1 from processor 1, move it to the list of ready threads, and then resume thread 5 by moving it from the ready list to run on processor 1.

Switching between threads is transparent to code being executed by the threads. Remember that the point of the abstraction is to make each thread appear to be a single stream of execution, so a programmer should only have to worry about the sequence of instruction within a thread and not have to worry about where that sequence may be (temporarily) suspended to let another thread run.

Threads thus provide an execution model in which *each thread runs on a dedicated virtual processor with unpredictable and variable speed*. From the point of view of a thread’s code, each instruction appears to execute immediately after the preceding instruction. However, the scheduler may suspend a thread between one instruction and the next and resume running it later. It is as if the thread was continually running on a processor that sometimes becomes very slow.

For example, Figure 4.3, illustrates a programmer’s view of a simple program and three (of many) possible ways that program might be executed, depending on what the scheduler does. From the point of view of the thread, other than the speed of execution, all of these ways are equivalent. Indeed, the thread would typically not even know which of these (or other) executions actually occurred.

Of course, a thread’s speed of execution and its interleavings with other threads will be affected by how it is scheduled. Figure 4.4, shows a handful of the many possible interleavings of a three thread program. Thread programmers should therefore not make any assumptions about the relative speed that different threads execute.

Why “unpredictable speed”? It may seem strange to assume that a thread’s virtual processor runs at a completely unpredictable speed and to assume that any interleavings with other threads are possible. Surely some interleavings are more likely than others?

The reason the threads programming model adopts this assumption is to guide programmers when reasoning about correctness. As Chapter 5 describes, rather than assuming that one thread at the same speed as another (or faster or slower) and trying to write programs that coordinate threads based on their rel-

Cooperative multithreading

Although most thread systems include a scheduler that can—at least in principle—run any thread at any time, some systems provide the abstraction of *cooperative threads*. In these systems, a thread runs without interruption until it explicitly relinquishes control of the processor to another thread. An advantage of cooperative multithreading is increased control over the interleavings among threads. For example, in most cooperative multithreading systems, only one thread runs at a time, so while a thread is running, no other thread can run and affect the system's state.

Unfortunately, cooperative multithreading has significant disadvantages. For example, a long-running thread can monopolize the processor, starving other threads and making the system's user interface sluggish or nonresponsive. Additionally, modern multiprocessor machines run multiple threads at a time, so one would still have to reason about the interleavings of threads even if cooperative multithreading were used. Thus, although cooperative multithreading was used in some significant systems in the past, including early versions of Apple's MacOS operating system, cooperative multithreading is less often used today.

The alternative we describe in the main body is sometimes called *preemptive multithreading* since a running thread can be preempted without explicitly relinquishing the processor. In the rest of this book, when we talk about multithreading, we are talking about preemptive multithreading unless we explicitly state otherwise.

Programmer's View	Possible Execution #1	Possible Execution #2	Possible Execution #3
.	.	.	.
.	.	.	.
$x = x + 1;$	$x = x + 1;$	$x = x + 1$	$x = x + 1$
$y = y + x;$	$y = y + x;$	$y = y + x$
$z = x + 5y;$	$z = x + 5y;$	thread is suspended other thread(s) run thread is resumed thread is suspended other thread(s) run thread is resumed
.
.
		$y = y + x$	
		$z = x + 5y$	$z = x + 5y$

Figure 4.3: Threads virtualize the underlying hardware's fixed number of processors to allow programmers to use any number of threads.

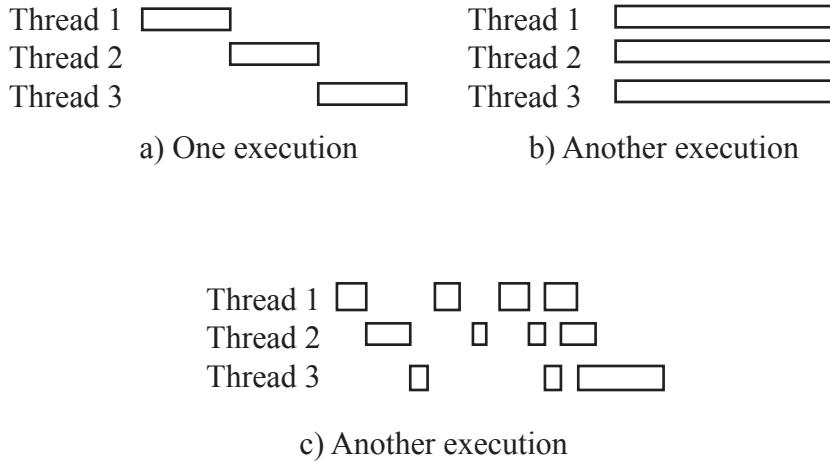


Figure 4.4: A handful of the many possible ways that three threads might be interleaved at runtime. Threads runs at unpredictable speeds due to many factors including policy decisions by the scheduler, the number of available processors, the physical characteristics of the processors, other programs competing for resources, and the arrival of I/O events.

ative speeds of execution, multi-threaded programs use explicit synchronization to control how threads interact with one another.

The physical reality is that the relative execution speeds of different threads are significantly affected by many factors outside their control. For example, on a modern processor, accessing memory can stall a processor for hundreds or thousands of cycles if there is a cache miss. Other factors include how frequently the scheduler preempts the thread, how many physical processors are present on a machine, how large the caches are, how fast the memory is, how the energy-saving firmware adjusts the processors' clock speeds, what network messages arrive, or what input is received from the user. As a result, execution speeds for the different threads of a program are hard to predict, can vary on different hardware, and can even vary from run to run on the same hardware.

4.2 Simple API and example

Figure 4.5 shows our simple thread API called `sthreads` (“simple threads.”) `Sthreads` is based on the Posix standard `pthreads` API, but it simplifies things by omitting some options and error handling. Although the `sthreads` API is based

```
void sthread_create(thread, func, arg)
```

Create a new thread, storing information about it in `thread`. The thread will execute the function `func`, which will be called with the argument `arg`.

```
void sthread_yield()
```

The calling thread voluntarily gives up the processor to let some other thread(s) run. The scheduler can resume running the calling thread whenever it chooses to do so.

```
int sthread_join(thread)
```

Wait for the specified thread `thread` to finish if it has not already done so; then return the value passed to `sthread_exit()` by the specified thread `thread`. Note that `sthread_join()` may be called only once for each `thread`.

```
void sthread_exit(ret)
```

Finish the current thread. Store the the value `ret` in the current thread's data structure and, if another thread is already waiting in a call to `sthread_join()`, then wake it up so that it can run and return this value.

Figure 4.5: Simple API for using threads called sthreads ("simple threads.")
If you want to run these examples or write your own, the sthread library code is available at `sthread.h` and `sthread.c`.

on Posix, most other threads packages are quite similar; if you understand how to program with sthreads, you will find it easy to write code with most standard APIs for threads.

4.2.1 A simple multi-threaded program

Figure 4.6 shows a simple multi-threaded program written in 'C' using sthreads.

The `main()` function uses `sthread_create()` to create 10 threads. The interesting arguments are the second and third.

- The second argument, `go`, is a pointer to the function `go()`, which is the function in which the newly-created thread should begin execution.
- The third argument, `ii`, is the argument that should be passed to that function.

Thus, `sthread_create()` will initialize the `ii`th thread's state so that it is prepared to call the function `go()` with the argument `ii`.

```

#include <stdio.h>
#include "sthread.h"

static void go(int n);

#define NTHREADS 10
static sthread_t threads[NTHREADS];

int main(int argc, char **argv)
{
    int ii;

    for(ii = 0; ii < NTHREADS; ii++){
        sthread_create(&(threads[ii]), &go, ii);
    }
    for(ii = 0; ii < NTHREADS; ii++){
        long ret = sthread_join(threads[ii]);
        printf("Thread %d returned %ld\n", ii, ret);
    }
    printf("Main thread done.\n");
    return 0;
}

void go(int n)
{
    printf("Hello from thread %d\n", n);
    sthread_exit(100 + n);
    // Not reached
}

```

Figure 4.6: Simple multi-threaded program.

When the scheduler runs the *i*ith thread, the thread runs the function `go()` with the value *ii* as an argument and prints `Hello from thread ii\n`. The thread then returns a value (*ii* + 100) by calling `sthread_exit(ii+100)`. This call stores the specified value in a field in the `sthread_t` object so that it can be retrieved by the initial thread's call to `sthread_join()`.

The `main()` function uses `sthread_join()` to wait for each of the threads it created to finish. As each thread finishes, code in `main()` reads the thread's exit value and prints it.

Figure 4.7 shows the output of one possible run of this program. Of course, other runs may give different interleavings of the output lines.

```
bash-3.2$ ./threadHello
Hello from thread 0
Hello from thread 1
Thread 0 returned 100
Hello from thread 3
Hello from thread 4
Thread 1 returned 101
Hello from thread 5
Hello from thread 2
Hello from thread 6
Hello from thread 8
Hello from thread 7
Hello from thread 9
Thread 2 returned 102
Thread 3 returned 103
Thread 4 returned 104
Thread 5 returned 105
Thread 6 returned 106
Thread 7 returned 107
Thread 8 returned 108
Thread 9 returned 109
Main thread done.
```

Figure 4.7: Output of one possible run of the program in Figure 4.6.

Exercises To complete the following problems, download the `sthread` library from `sthread.h`, and `sthread.c`. The comment at the top of `threadHello.c` explains how to compile and run a program that uses this library.

1. Download this example code `threadHello.c`, compile it, and run it several times. What do you get if you run it? Do you get the same thing if you run it multiple times? What if you are also running some other demanding processes (e.g., compiling a big program, playing a Flash game on a website, or watching streaming video) when you run this program?
2. For the `threadHello` program in Figure ??, what is the *maximum* number of threads that could exist while the program is running? (*Be careful.*)
3. For the `threadHello` program in Figure ??, suppose that we delete the second `for` loop so that the main routine simply creates `NTHREADS` threads and then prints “Main thread done.” What are the possible outputs of the program now. **Hint:** Fewer than `NTHREADS+1` lines may be printed in some runs. Why?
4. How expensive are threads? Write a program that times how long it takes to create and then join 1000 threads, where each thread simply calls `sthread_exit(0)` as soon as it starts running.
5. Write a program that has two threads. The first thread a simple loop that continuously increments a counter and prints a period (“.”) whenever

the value of that counter is divisible by 10,000,000. The second thread repeatedly waits for the user to input a line of text and then prints “Thank you for your input.” On your system, does the operating system do a good job of making sure that the first thread makes rapid progress and that the second thread responds quickly?

6. Write a program that performs a parallel matrix multiply...

4.3 Thread internals

Each thread represents a sequential stream of execution, and the operating system provides the illusion that each thread runs on its own virtual processor by transparently suspending and resuming threads.

To understand how the operating system implements the threads abstraction, we must define the state needed to represent a thread. Then, we can understand a thread’s lifecycle—how the operating system can create, start, stop, and delete threads to provide the abstraction.

4.3.1 Thread control block (TCB) and per-thread state

The operating system needs a data structure to represent a thread’s state; a thread is like any other object in this regard. The data structure that hold the thread’s state is called the *thread control block (TCB)*. For every thread the operating system creates, it will create one TCB.

Definition: **thread control block (TCB)**

A thread’s TCB holds two types of information:

- The state of the computation being performed by the thread
- Metadata about the thread that is used to manage the thread

Per-thread computation state. Each thread represents its own sequentially-executed computation, so to create multiple threads, we must allocate per-thread state to represent the current state of each thread’s computation. Each thread’s TCB therefore contains two elements of per-thread computation state:

1. Stack.

A thread’s stack is the same as the stack for a single-threaded computation—it stores information needed by the nested procedures the thread is currently running. For example, if a thread calls `foo()`, `foo()` calls `bar()`,

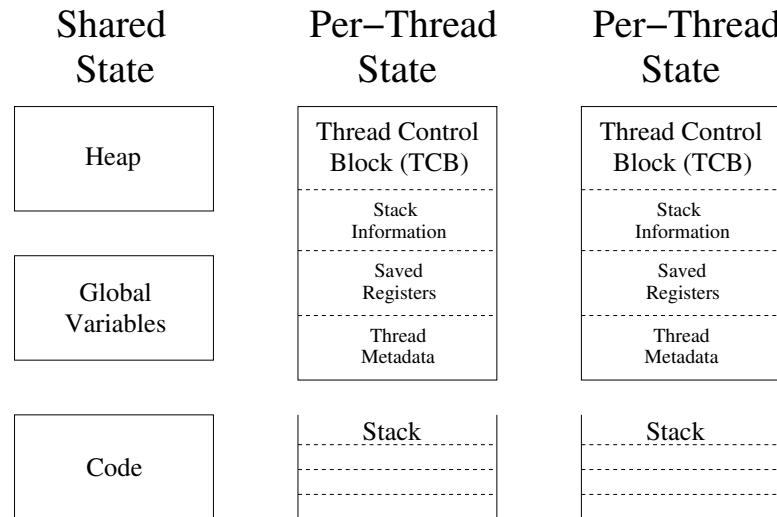


Figure 4.8: A multi-threaded process or operating system's state is divided into *per thread state* and *shared state*. The thread control block is the object that stores per-thread state. The per-thread state includes the state needed to represent the thread's computation (e.g., the processor registers, the stack, and thread-local variables.) The per-thread state also included metadata—data about the thread used by the operating system in managing the thread (e.g., the thread's ID, scheduling priority, owner, and resource consumption.) In addition to per-thread state, a multi-threaded process (or the operating system kernel) includes shared state, shared by all threads running in that process (or kernel). For example, this shared state includes the program's code, global static variables, and the heap.

and `bar()` calls `bas()`, then the stack would contain a *stack frame* for each of these three procedures, and each stack frame might contain the local variables used by the procedure, the parameters the procedure was called with, and the return address to jump to when the procedure completes.

Because at any given time, each thread can be in a different state in its sequential computation—it can be in a different place in a different procedure called with different arguments from a different nesting of enclosing procedures—each thread needs its own stack. Typically, when a new thread is created, a new stack is allocated for it, and a pointer to that stack is stored in the thread's TCB.

2. **Copy of processor registers.** A processor's registers include not only general purpose registers for storing values for ongoing computations but also special purpose registers like the instruction pointer and stack pointer.

Because the operating system needs to be able to suspend a thread, run another thread, and then later resume the original thread, the operating

Other per-thread state: Thread-local variables

In addition to the per-thread state that corresponds to execution state in the single-threaded case, some systems include additional *thread-local variables*. These variables are similar to global variables in that their scope spans different procedures, but they differ in that each thread has its own copy of these variables.

Example: Errno. In UNIX programs a variable called `errno` stores information about any errors that occurred in the most recently executed system call. In a multi-threaded program, multiple threads can perform system calls concurrently. Thus, to retrieve the status of the current thread's most recent library call, `errno` is typically a macro that maps to a thread-local variable.

Example: Heap internals. Although a program's heap is logically shared—it is OK for one thread to allocate an object on the heap and then pass a pointer to that object to another thread—for performance reasons heaps may internally subdivide their space into per-thread regions. The advantage of subdividing the heap this way is that multiple threads can each be allocating objects at the same time without interfering with one another, and cache hit rates may be better. For this to work, each subdivision of the heap has some thread-local variables that keep track of what parts are allocated, what parts are free, and so on. Then, the code that allocates new memory (e.g., `malloc()` and `new()`) are written to use these thread-local data structures.

Thread-local variables are useful, but for simplicity, the rest of our discussion focuses on just the TCB, registers, and stack as the core pieces of per-thread state.

system needs somewhere to store a thread's registers when that thread is not actively running on a processor. So, the TCB contains fields in which the operating system can store a copy of all of the processor's registers.

Per-thread metadata. The TCB also includes include *per-thread metadata*—information about each thread. This information is used by the operating system to manage the thread. For example, each thread might have a thread ID, scheduling priority, and any other information the operating system wants to remember about the thread.

Shared state. In addition to per-thread state that is allocated for each thread, as Figure 4.8 illustrates, other state is *shared* by different threads in a multi-threaded process or multi-threaded operating system.

In particular, the program *code* is shared by all of the threads in a process. Additionally, statically-allocated *global variables* and dynamically-allocated *heap variables* can store information that is accessible to all threads.

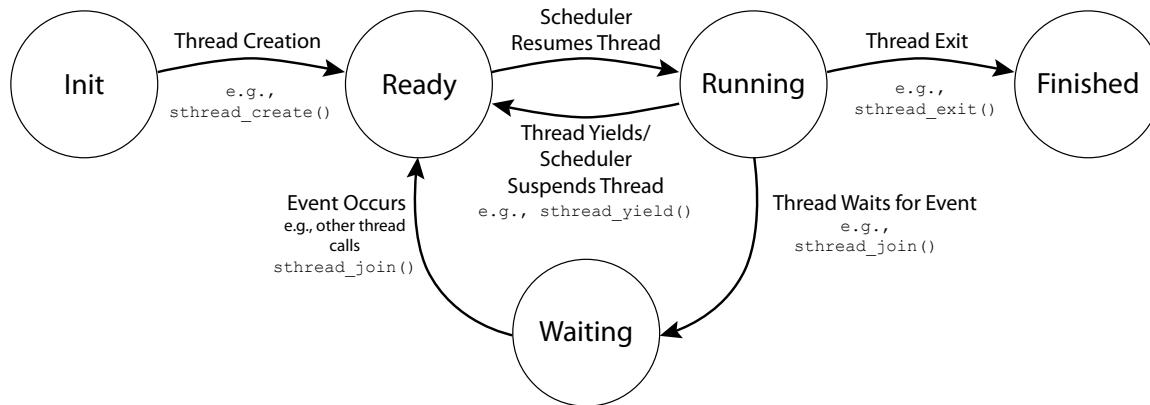


Figure 4.9: The states occupied by a thread during its life cycle.

Warning. Note that although there is an important logical division between per-thread state and shared state, nothing in the operating system typically *enforces* this division, and nothing protects a program from bugs when one thread accesses another thread's (conceptually private) per thread state.

So, writing to a bad pointer in one thread might corrupt the stack of another, or a careless programmer might pass a pointer to a local variable on one thread's stack to another thread, giving the second thread a pointer to a stack location whose contents may change as the first thread calls and returns from various procedures. Because they can depend on the specific interleavings of the threads' executions, such bugs can be diabolically hard to find.

It is therefore important when writing multi-threaded programs to know what variables are designed to be shared across threads (global variables, objects on the heap) and what variables are designed to be private (local/automatic variables) in order to avoid unexpected behaviors.

4.3.2 Thread life cycle

Figure 4.9 shows the states occupied by a thread during its lifetime.

Init. Thread creation puts a thread into its initial INIT state and allocates and initializes per-thread data structures. Once that is done, the thread creation code puts the thread into the READY state by adding the thread to a list of READY threads called the *ready list*.

Ready. A thread in the READY state is available to be run but it is not currently running. Its TCB is on the ready list and the values of its registers are stored in the thread control block. At any time, the scheduler can cause

The idle thread.

If a system has k processors, then there will normally be exactly k RUNNING threads; the operating system typically keeps a low priority *idle thread* per processor and runs it if there is nothing else to run.

In old machines, the idle thread would spin in a tight loop doing nothing.

Today, to save power, the idle thread is typically a loop that on each iteration puts the processor into a low-power sleep mode in which the processor stops executing instructions until a hardware interrupt occurs. Then, when a hardware interrupt occurs, the processor wakes up and handles it in the normal way—saving the state of the currently running thread (the idle thread) and running the handler. After running the handler, if a thread other than the idle thread is READY, the scheduler runs that thread next; otherwise, the idle thread resumes execution, putting the processor to sleep again.

a a thread to transition from the READY to RUNNING by copying the thread's register values from its TCB to a processor's registers.

Running. A thread in the RUNNING state is running on a processor. When a thread is RUNNING, its register values are stored on a processor rather than in the TCB data structure. A RUNNING thread can transition to the READY state in two ways.

- The scheduler can preempt a running thread and move it to the READY state at any time by saving the thread's registers to its TCB and switching the processor to running the next thread on the ready list.
- A running thread can voluntarily relinquish the processor go from RUNNING to READY by calling `yield()` (e.g., `sthread_yield()` in the `sthreads` library.)

Notice that a thread can transition between READY and RUNNING and back many times while it runs. Since the operating system saves and restores the thread's registers exactly, only the speed of the the thread's execution is affected by these transitions.

Waiting. A thread in the WAITING state is waiting for some event. Whereas a thread in the READY stage is eligible to be moved the RUNNING state by the scheduler, a thread in the WAITING state is not eligible to be run until some action by another thread moves it from the WAITING state to the READY state.

Example: A WAITING thread. We saw an example of this earlier in the multi-threaded program shown in Figure 4.6 on page 147. After creating a number of

State of Thread	Location of Thread Control Block (TCB)	Location of Registers
INIT	Being Created	TCB
READY	Ready List	TCB
RUNNING	Running List	Processor
WAITING	Synchronization Variable's Waiting List	TCB
FINISHED	Finished List then Deleted	TCB

Figure 4.10: Location of thread's per-thread state for different life cycle stages.

threads, the main thread receives the exit value for each of the other threads thread, so it calls `sthread_join()` once for each of the child threads. If the child thread is not done yet, the main thread waits for it to finish so it can learn the exit value the child passed to `sthread_exit()`. In this case, the main thread goes from RUNNING to WAITING until the child's exit value is available.

While a thread is waiting for an event, it cannot make progress, so it is not useful to run it. Therefore, rather than continuing to run the thread or storing the TCB on the scheduler's ready list, the TCB is stored on the *waiting list* of some *synchronization variable* associated with the event. Then, when the event occurs, the TCB can be moved from the synchronization variable's waiting list to the scheduler's ready list, thus transitioning the thread from the WAITING state to the READY state. We will describe synchronization variables in the next chapter.

Finished. Finally, a thread in the FINISHED state will never run again. The system can free some or all of its state for other uses, though it may keep some remnants of the thread in the FINISHED state for some time by putting the TCB on a *finished list*. For example, our `sthread_exit()` call allows a thread to pass its exit value to its parent thread via `sthread_join()`. Eventually, when a thread's state is no longer needed (e.g., after its exit value has been read by a `join()` call), the thread's state can be deleted and reclaimed by the system.

Follow the bouncing TCB. As Figure 4.10 summarizes, one way to understand these stages is to think about where a thread's TCB and registers are stored. For example, all of the threads in the READY state have their TCBs on the ready list and their registers in the TCB, all threads in the RUNNING state have their TCBs on the running list and their registers on the processors, and all threads in the WAITING state have their TCBs on various synchronization variables' waiting lists.

Exercises

7. For the `threadHello` program in Figure ??, when `pthread_join()` returns for thread `ii`, in which of the states shown in Figure 4.9 is thread `ii`?
8. For the `threadHello` program in Figure ?? the procedure `go()` has the parameter `np` and the local variable `n`. Are these variables *per thread* or *shared* state? Where does the compiler store these variables' states?
9. For the `threadHello` program in Figure ?? the procedure `main()` has local variables such as `ii`, `err`, and `status`. Are these variables *per thread* or *shared* state? Where does the compiler store these variables' states?
10. In the sidebar on page 151, we describe *thread-local variables*, which are another piece of per-thread state present in many thread systems.

Describe how you would implement thread-local variables. Each thread should have an array of 1024 pointers to its thread-local variables.

- What would you add to the TCB?
- How would you change the thread creation procedure? (For simplicity, assume that when a thread is created, all 1024 entries should be initialized to NULL.)
- How would a running thread allocate a new thread-local variable?
- In your design, how would a running thread access a particular thread-local variable?

4.4 Implementation details

In the discussion above, we sketched the basic operation of threads. To make things concrete, we now describe how to implement threads in more detail.

Types of threads. Operating systems use threads internally—the operating system kernel can be multi-threaded—and they also provide the abstraction of threads to user-level processes so that processes can be multi-threaded.

The implementation of these two cases is almost identical. For simplicity we initially focus on how to implement *in-kernel threads*. This case is the simplest one because all of the scheduling and thread-switching actions occur in one place—the kernel. We will then discuss the small changes needed to extend the threads abstraction to support multi-threaded processes.

```

// func is a pointer to a procedure we want the thread to run
// arg is the argument to be passed to that procedure
void
thread_create(sthread_t *thrd, void (*func)(int), int arg)
{
    // Allocate TCB and stack
    TCB *tcb = new TCB();
    thrd->tcb = tcb;
    tcb->stack_size = INITIAL_STACK_SIZE;
    tcb->stack_base = new Stack(INITIAL_STACK_SIZE);

    // Initialize register values
    tcb->setStackPointer(stack);
    tcb->setInstructionPtr(stub); // Don't call func() directly
    tcb->setArg0Register(func); // Tell stub to call func()
    tcb->setArg1Register(arg); // Tell stub to use arg when it calls func()
    ...

    // Put thread in READY state
    readyList->add(tcb); // Put tcb on ready list
}

void
stub(void (*func)(int), int arg)
{
    (*func)(arg); // Execute the function func()
    thread_exit(0); // If func() doesn't call exit, call it here
}

```

Figure 4.11: Pseudo-code for thread creation. For convenience, we assume that the stack grows from low addresses to high ones and that arguments to functions are passed in registers. The code would be somewhat different for the x86 architecture, where the stack grows down and arguments are passed on the stack.

4.4.1 Creating a thread

Figure 4.11 shows the pseudo-code to allocate a new thread. Since we have to say what we want the thread to do, when we call `sthread_create()` we hand it a pointer to a procedure `func()` and an argument that we would like to pass to that procedure `arg`. Then, when the thread we create runs, it will execute `func(arg)`.

Allocating per-thread state. To allocate a thread, we must first allocate its per-thread state. As the pseudo-code illustrates, the thread constructor therefore allocates a *thread control block* (TCB) and *stack*. The TCB is just a data structure with fields for the state we want to be able to maintain for a thread. The stack is just an area of memory like any other data structure allocated in memory.

Note that we must choose some size for the allocated stack. Some implementations allocate a fixed-sized stack, and it is a bug for a thread's stack to overflow the fixed-sized stack. Other systems allocate an initial stack of some

size and dynamically grow it if needed.

Initializing per-thread state. When we initialize the thread control block, we need to set the new thread's registers to some sensible initial values. In the pseudocode, we first set the thread's stack pointer to the base of the newly allocated stack. Then, we initialize the thread's instruction pointer and argument registers so that when it runs it will call `func(arg)`.

Note that rather than having the thread call `func(arg)` directly, we initialize the thread's state so that the thread will first call the function `stub()` with two arguments: `func`, the function to run, and `arg`, the arguments to that function. The function `stub()` then calls `func(arg)` to run the function specified by the caller. We add this extra step so that if the `func()` procedure returns (rather than calling `thread_exit()` when it is done), it has somewhere to return to. If `func()` does return to `stub()`, `stub()` calls `thread_exit()` itself to finish this thread.

4.4.2 Deleting a thread

To delete a thread, we need to (1) remove it from the ready list so that it will never run again and (2) free the per-thread state we allocated for it.

There is one subtlety: if a thread removes itself from the ready list and frees its own per-thread state, then the constructs we've described will break. For example, if a thread removes itself from the ready list but an interrupt occurs before the thread finishes deallocating its state, we have a memory leak since the thread will never resume to finish deallocating its state. Worse, suppose that a thread frees its own state; what does it do then? How can it continue running the rest of the code in `sthread_exit()` if it does not have a stack? What happens if an interrupt occurs after the running thread's state has been deallocated? If the context switch code tries to save the current thread's state to its TCB, it will be writing to freed memory, possibly corrupting memory and causing a subtle bug.

There is a simple fix. A thread does not delete its own state. Instead, a thread transitions to the FINISHED state by moving its TCB from the ready list to a list of *finished* threads the scheduler should never run. Then, any time the thread library code runs (e.g., when some other thread calls `yield()` or when an interrupt occurs), the thread library can call a method to free the state of any threads on the finished list.

Thus, `sthread_delete()` merely moves the current thread to the finished list and then yields to some thread on the ready list. Later on, it is safe for some *other* thread to free the state of the threads on the finished list.

4.4.3 Thread context switch

When the kernel has multiple threads, we need a mechanism to switch which threads are RUNNING and which are READY.

Definition: **thread context switch**

A *thread context switch* suspends execution of the currently running thread and resumes execution of some other thread. This is done by copying the currently running thread's registers from the processor to the thread's TCB and then copying the other thread's registers from that thread's TCB to the processor.

The more things change, the more they stay the same. We've already seen in Chapter 2 how an interrupt, an exception, or a system call trap can cause a *mode switch* that suspends a running process, runs a kernel handler, and returns to running the original process.

Recall that to implement a mode switch, the hardware saves some of the registers of the currently running process, switches to kernel mode, and starts running the handler. Then the handler software saves the remaining registers, handles the event, and finally restores the suspended process's registers and mode to resume running the suspended process exactly where it left off.

The mechanism for switching between threads in the kernel is almost identical. For example, if an interrupt or exception occurs: the hardware saves some of the registers of the currently running process and starts running a handler; the handler then saves the remaining registers, and begins running the main body of the handler.

Then, when the handler is done running, instead of resuming the interrupted process, it can invoke the scheduler to choose a different thread to run rather than restoring the thread that was interrupted. This is easy to do: we have already saved the state of the interrupted thread and we have a ready list of other threads whose state has already been saved. We can copy the interrupted thread's state to its TCB and copy the state of some ready thread from its TCB to the registers; then our thread switch is done.

The devil is in the details. There are some small differences between the two cases, but they amount to implementation details rather than changes in the big picture. We describe these details below by discussing

- What triggers a context switch?
- How does context switch work?

The rest of this section discusses these issues.

Note that we defer discussing one issue

Separating mechanism from policy

Separating mechanism from policy is a useful and widely-applied principle in operating system design. When mechanism and policy are cleanly separated, it is easier to introduce new policies to optimize a system for a new workload or new technology.

For example, the thread context switch abstraction cleanly separates mechanism (how to switch between threads) from policy (which thread to run) so that the mechanism works no matter what policy is used. Then, some systems can elect to do something simple (e.g., FIFO scheduling), while other systems can optimize scheduling to meet their goals (e.g., a periodic scheduler to smoothly run real-time multimedia streams for a media device, a round-robin scheduler to balance responsiveness and throughput for a server, or a priority scheduler that devotes most resources to the visible application on a smartphone.)

We will see this principle many times in this book. For example, thread synchronization mechanism are also designed to work regardless of the scheduling policy, file metadata mechanisms for locating a file's blocks are designed to work regardless of the policy for choosing where to place the file's blocks on disk, and page translation mechanisms for mapping virtual page addresses to physical page addresses work regardless of which ranges of virtual addresses a process uses and which physical pages the operating system assigns to it.

- Which READY thread should the scheduler choose to run next?

The *mechanisms* we discuss work regardless of what *policy* the scheduler uses when choosing threads.

As for the *policy* for choosing which READY to run next, there are trade-offs among different scheduling policies, and different operating systems use different approaches. We therefore defer discussing scheduling policy to Chapter 7 where we can discuss these policy trade-offs in depth.

For now, you can assume that the scheduler's ready list is a simple FIFO list; most schedulers do something more sophisticated—they may prioritize some threads over others, optimize for throughput or responsiveness, or run ensure that multimedia threads run at a regular interval—but nothing we talk about here depends on these details.

What triggers a kernel thread context switch?

When a thread in the kernel is running, two things can cause a thread context switch.

First, the thread may *call a thread library function* that triggers a context switch. For example, many thread libraries provide a `thread_yield()` call

that allows the currently running thread to voluntarily give up the processor and context switch the processor to some other thread that is ready to run. Similarly, the `thread_join()` and `thread_exit()` calls can suspend execution of the current thread and start running a different one.

Second, an *interrupt* or *exception* may invoke an interrupt handler, which must save the state of the running thread, execute the handler's code, and switch to some thread that is ready to run (or it may just restore the state and resume running the thread that was just interrupted.)

For example, many thread libraries switch threads when a timer interrupt occurs. In particular, thread libraries typically want to ensure that no thread can monopolize the processor, so they arrange for a hardware timer to periodically (e.g., every 10ms) cause a *timer interrupt*. The handler for the timer interrupt saves the state of the running thread, chooses another thread to run, and runs that thread by restoring its state to the processor.

Other hardware events (e.g., a keyboard key is pressed, a network packet arrives, or a disk operation completes) also invoke interrupt handlers. In all cases, these handlers save the state of the currently running thread so that it can be restored later. They then execute the handler code, and when the handler is done, they restore the state of some ready thread. If the interrupt caused a high priority thread to become ready, then the scheduler will set things up so that the interrupt returns to that thread instead of the interrupted one.

How does a kernel thread context switch work?

Regardless of whether a switch between kernel threads is triggered via an interrupt or an explicit call to the threads library, what needs to be done is conceptually simple:

1. Copy the running thread's registers from the processor to the thread's TCB
2. Copy a ready thread's registers from the thread's TCB to the processor

The implementation details differ slightly depending on whether the thread switch is caused by an interrupt or a explicit library call by the thread. Still, the cases are more the same than different—we want to make sure that all of the threads on the ready list have their state saved in the same way so that we can restore them in the same way.

Hardware-triggered thread switch. We saw what happens when an interrupt, exception, or trap interrupts a running user-level process in Chapter ??: hardware and software work together to save the state of the interrupted process, run the kernel's handler, and restore the state of the interrupted process.

The mechanism is almost identical when an interrupt or trap triggers a thread switch among threads in the kernel. We tweak the three steps described in Chapter ?? for handling a context switch from a user-level process to a kernel handler as follows (changes are written in italics):

1. **Save the state.** Save the currently running *thread's* registers so that the handler can run code without disrupting the interrupted *thread*.

Recall that this is done with a combination of hardware saving some state when the interrupt or exception occurs and software saving the rest of the state when the handler runs.

2. **Run the kernel's handler.** Run the kernel's handler code to handle the interrupt or exception.

Since we are already in kernel mode, we do not need to change from user to kernel mode in this step.

We also do not need to change the stack pointer to the base of the kernel's exception stack. Instead, we can just push and pop saved state or handler variables onto the current stack starting from the current stack pointer.

3. **Restore the state.** Restore the *next ready thread's* registers so that it can resume running where it left off.

In short, comparing this approach to what happens on a mode switch from a user-level process, the only significant changes are (1) we do not need to switch modes (and therefore do not need to switch stacks) and (2) we can resume any thread on the ready list rather than always resuming the thread or process that was just suspended.

Library-call-triggered thread switch. When a thread calls a library function such as `thread_yield()` that triggers a thread switch, the steps are similar to what an interrupt or exception does. We need to

1. **Save the state.** Save the currently running thread's registers so that the kernel can later resume this thread.

Here there is no hardware event; the running thread just calls the kernel handler with a regular procedure call. So, the library procedure must save all of the state on its own.

That said, the library software should save the state to the thread's TCB using the same format as used in the interrupt- or exception-triggered case so that we can restore threads the same way regardless of what caused the thread to be suspended.

2. **Run the called library procedure.** Run the code to handle the library call.

```

void thread_yield()
{
    // Make stack look similar to the interrupt case (e.g., Fig. ??)
    push eflags; // Push processor execution flags onto stack
    push LABEL; // Push address of LABEL instruction (below)
                 // instead of current instruction pointer. In x86
                 // the pushed address needs to include both the
                 // code segment (cs) and program counter (eip).
    pushad;      // Push general purpose registers

    // Copy the current thread's saved state to TCB data structure
    memcpy(&(currentThread->TCB.savedRegisters), stackPointer, STATE_SIZE);

    // Choose another TCB from the ready list
    chosenTCB = readyList->getNextThread();

    //
    // Restore state of the chosen thread
    //
    // First, set stack pointer to point to area of TCB data structure
    // that stores the saved state. Then, restore the rest of the
    // state as if we were just popping it off the stack.
    stackPointer = &(chosenTCB->savedRegisters);
    popad;        // Pop other thread's general purpose registers
    iret;         // Pop other thread's instruction pointer and execution flags

    // This line is never executed!
    assert(0);

    // When calling thread is resumed, it will start here
LABEL:
    return;
}

```

Figure 4.12: Pseudo-code for `thread_yield()` for in-kernel threads on x86 architecture.

3. **Restore the state.** Restore the next ready thread's registers so that it can resume running where it left off.

As you can see, these steps are quite similar to the steps for an interrupt- and exception-triggered thread context switch. This is no accident!

Example: `thread_yield()` on x86. To make this software-based approach concrete, Figure 4.12 shows pseudo-code for a simple implementation of `thread_yield()` for in-kernel threads for the x86 hardware architecture. A thread calls `thread_yield()` to voluntarily relinquish the processor to another thread. The calling thread's registers are copied to its TCB on the ready queue, and that thread resumes running later, when the scheduler chooses it.

In this code, the calling thread first saves its state to its stack. In our pseudo-code we take care to ensure the stack we construct here looks like the one constructed when an interrupt triggers the context switch (Figure ?? on page ??).

So, we first push the x86 `eflags` register, which includes the mode bit (user v. kernel) and a flag indicating whether interrupts are masked or enabled.

Then, we push an instruction pointer. Notice that—perhaps surprisingly—the instruction pointer we push is not the address of the current instruction. Instead, we save the address of the instruction at `LABEL`, near the end of the `thread_yield()` procedure. We need to do this to avoid an endless loop. In particular, if we instead stored a pointer to the current instruction, then later, when this thread is restored, it would continue where it left off (e.g., it would finish saving its state to the stack, copy its saved state to its TCB, and run another thread.)

Finally, we save the remaining registers to the stack using the x86 `pushad` instruction.

Once the calling thread saves its state to its stack, it copies its saved state to its TCB, chooses another thread to run, and runs it by restoring its state in three steps: (1) change the stack pointer to point to the other thread's saved registers; (2) pop the other thread's general purpose registers; and (3) pop the other thread's instruction pointer and execution flags. After step (3), the current thread is no longer executing, and the `assert(0)` in the pseudo-code is never reached.

When the original thread is eventually restored, its stack, stack pointer, and all registers except the instruction pointer are the same as they were when the `pushad` call saved the registers. The instruction pointer is now `LABEL`, and the first thing the thread does upon being resumed is to execute the `return`. In essence, `thread_yield()` appears to the caller as an empty procedure that does nothing but immediately return. Of course, behind the scenes a lot is happening.

Processor’s point of view. From the processor’s point of view, one instruction follows the next, but now the instructions from different threads are interleaved (as they must be if we are to support multiplexing!)

This interleaving can seem a bit unusual—`thread_yield()` deliberately violates the procedure call conventions compilers normally follow by manipulating the stack and program counter to switch between threads. For example, Figure 4.13 shows the interleaving when two threads each execute a simple endless loop:

```
while(1){
    thread_yield();
}
```

One way to view Figure 4.13 is that from the processor’s point of view—because of the way a context switch manipulates the registers—`thread_yield()` is called by one thread but returns in a different thread.

But, once the operating system has the needed mechanisms to switch between threads, the threads, themselves, get to ignore this complexity. From their point of view, they each just run this loop on their own (variable-speed) virtual processor.

One small difference.

You may notice that in Chapter 2, for a mode switch the x86 hardware saved not just the instruction pointer and `eflags` register but also the the *stack pointer* of the interrupted process before starting the handler. In the mode switch case, the hardware changed the stack pointer to the kernel's exception stack, so it must save the original stack pointer.

In contrast, when we switch from a kernel thread to a kernel handler—as we do for a thread context switch initiated by either a library call or an interrupt—the hardware does not switch stacks. Instead, the handler runs on the current stack not a separate exception stack. Therefore, the hardware does not need to save the original stack pointer; the handler just saves the stack pointer with the other registers as part of the `pushad` instruction.

Thus, x86 hardware thus works a bit differently when switching between a kernel thread and a kernel handler than when doing a mode switch.

- **Entering the handler.** When an interrupt or exception occurs, if the processor detects that it is already in kernel mode (by inspecting the `eflags` register), it just pushes the instruction pointer and `eflags` registers (but not the stack pointer) onto the existing stack.

On the other hand, if the hardware detects that it is switching from user mode to kernel mode, then the processor also changes the stack pointer to the base of the exception stack and pushes the original stack pointer along with the instruction pointer and `eflags` registers onto the new stack.

- **Returning from the handler.** When the `iret` instruction is called, it inspects both the current `eflags` register and the value on the stack that will use to restore the earlier `eflags` register. If the mode bit is identical, then `iret` just pops the instruction pointer and `eflags` register and continues to use the current stack.

On the other hand, if the mode bit differs, then the `iret` instruction pops not only the instruction pointer and `eflags` register, but also the saved stack pointer, thus switching the processor's stack pointer to the saved one.

To be compatible with this x86 hardware behavior, our software implementation of `thread.yield` for in-kernel threads simulates the hardware case, saving only the instruction pointer and `eflags` register before calling `pushad` to save the general-purpose registers (including the stack pointer.) Now, `iret` will work properly whether the kernel thread it is resuming was suspended by a hardware event or a software call.

Logical View		
Thread 1	Thread 2	
<pre>while(1){ thread_yield() }</pre>	<pre>while(1){ thread_yield() }</pre>	
Physical Reality		
Thread 1's instructions	Thread 2's instructions	Processor's instructions
call thread_yield save state to stack save state to TCB choose another thread load other thread state	call thread_yield save state to stack save state to TCB choose another thread load other thread state	call thread_yield save state to stack save state to TCB choose another thread load other thread state
return thread_yield call thread_yield save state to stack save state to TCB choose another thread load other thread state	call thread_yield save state to stack save state to TCB choose another thread load other thread state	return thread_yield call thread_yield save state to stack save state to TCB choose another thread load other thread state
return thread_yield call thread_yield save state to stack save state to TCB choose another thread load other thread state	return thread_yield call thread_yield save state to stack save state to TCB choose another thread load other thread state	return thread_yield call thread_yield save state to stack save state to TCB choose another thread load other thread state
...

Figure 4.13: Interleaving of instructions when two threads loop and call `thread_yield()`.

4.4.4 Multi-threaded processes

So far, we have described how to implement multiple threads in a kernel. Threads are also a useful abstraction for applications, so modern operating systems support *multi-threaded processes*.

We will describe how operating systems provide the abstraction of multi-threaded processes in two steps.

- First, we will describe how single-threaded processes mesh with the mechanisms we've just described for implementing a multi-threaded operating system.
- Then, we will describe a few small additions needed to support multi-threaded processes.

A zero-thread kernel

Not only can we have a single-threaded kernel or a multi-threaded kernel, it is actually possible to have a kernel with no threads of its own—a zero-threaded kernel! In fact, this used to be quite common.

Consider the simple picture of an operating system we sketched in Chapter 2. Once the system has booted, initialized its device drivers, and started some user-level processes like a login shell, everything else the kernel does is event-driven—done in response to an interrupt, exception, or trap.

In a simple operating system like this, there is no need for a “kernel thread” or “kernel thread control block” to keep track of an ongoing computation. Instead, when an interrupt, trap, or exception occurs, the stack pointer gets set to the base of the exception stack, the instruction pointer gets set to the address of the handler. Then the handler executes and either returns immediately to the interrupted user-level process or suspends the user-level process and “returns” to some other user-level process. In either case, the next event (interrupt, trap, or exception) starts this process anew.

Multi-threaded kernel with single-threaded processes

Figure 4.14 illustrates two single-threaded user-level processes running on a multi-threaded kernel with three kernel threads. Notice that each user-level process includes the process’s thread. But, each process is more than just a thread because each process has its own address space—process 1 has its own view of memory, its own code, its own heap, and its own global variables that differ from those of process 2 (and differ from those of the kernel).

Because a process is more than just a thread, each process’s *process control block* (PCB) needs more information than a thread control block (TCB) for a kernel thread, and switching between processes—or between a kernel thread and a user-level process—needs to do a bit more work.

PCB v. TCB. The a kernel thread’s TCB and a user-level process’s PCB are similar, but it has some additional information. Like a TCB, a PCB must store the processor registers when the process’s thread is not running. In addition, the PCB has information about the process’s address space so that when we context switch from one process to another or between a process and the kernel, the right virtual memory mappings are used. We discuss virtual memory in Chapter ??.

With respect to concurrency and threads, the PCB and TCB each represents one thread, and the kernel’s ready list contains a mix of PCBs for processes and TCBs for kernel thread. When the scheduler chooses the next thread to run, it can pick either kind.

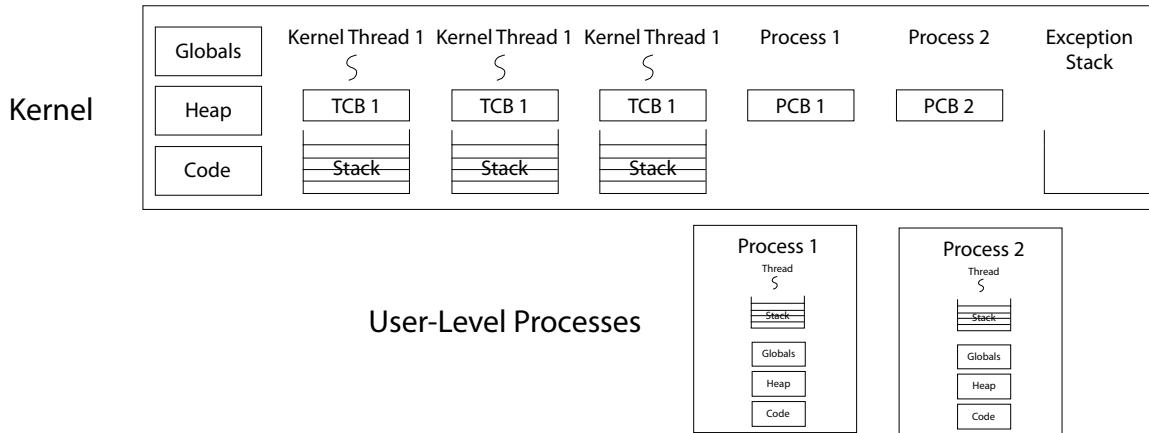


Figure 4.14: A multi-threaded kernel with 3 kernel threads and 2 single-threaded user-level processes.

Process switch v. thread switch. At a high level, thread switch is nearly identical whether we are switching between kernel threads or switching between a process's thread and a kernel thread. In all cases, we save the state of the currently running thread and restore the state of the next thread to run.

The difference is that when switching from a user-level thread, we need to switch from operating in user mode to kernel mode. Leaving aside the additional work needed to change the virtual memory mappings (which we'll discuss in Chapter ??), the mode change can also slightly change the low-level implementation details for saving and restoring thread state.

- **Hardware-triggered (interrupts and exceptions).** When an interrupt or exception causes a hardware-triggered thread switch, the hardware and software work together to save the state of the suspended thread. Exactly what state gets saved by the hardware may differ slightly depending on whether the interrupted thread is running in user mode or kernel mode, but these changes are relatively small implementation details. For example, we saw that for the x86 architecture, the hardware saves the stack pointer when it switches from user to kernel mode but not when it stays in kernel mode, and the `iret` instruction to resume a suspended thread has to account for this difference.
- **Software-triggered (library calls v. system calls).** When an in-kernel thread accesses the threads library to create or delete a thread or to suspend, resume, or switch threads it can use a simple procedure call, but when a user-level thread accesses the threads library to do these things, it needs to use a system call; because the PCBs are in the kernel's memory, a user-level process's threads must invoke kernel code to save their state.

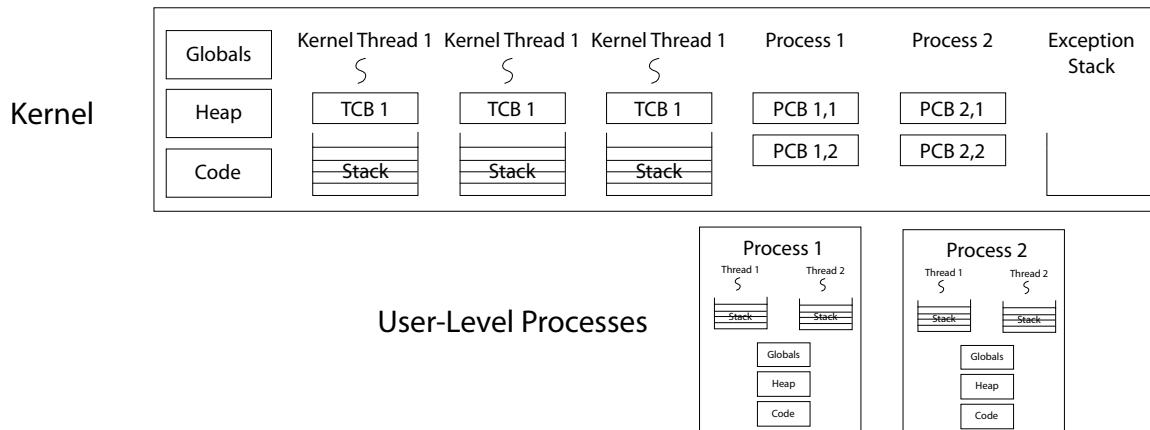


Figure 4.15: A multi-threaded kernel with 3 kernel threads and 2 user-level processes, each with 2 threads.

Multi-threaded kernel with multi-threaded processes

The basic infrastructure just described for running a mix of kernel threads and single-threaded user processes needs little change to run multi-threaded user processes such as the mix shown in Figure ???. The kernel's ready list includes kernel thread TCBs and one or more PCBs for each user-level process, and the thread context switch mechanisms work as described in the previous subsection to allow switching between kernel threads, between a kernel thread and a process thread, between threads from different processes, or between threads from the same process.

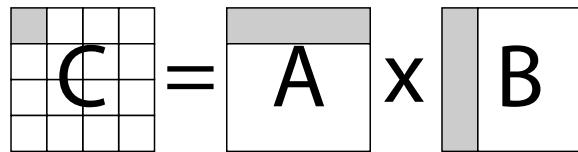
A process can now use system calls to create multiple threads. Notice that for each thread, the thread's PCB is created in the kernel, but the thread's stack is allocated in the process's memory. But, the process's threads share other state such as the process's code, heap, and globals.

A process can create multiple threads for the reasons listed at the start of the chapter.

1. **Program structure.** A program that needs to do multiple things at the same time can create a thread for each one and rely on the operating system to transparently switch among these threads so that each makes progress.
2. **Exploiting multiple processors.** If a program can divide its work into independent pieces, it can create a thread for each piece. If multiple processors are available—this depends on what hardware the program is running on and on what other kernel threads and other processes are

active—the operating system may spread this program’s threads across multiple processors, allowing it to finish its work more quickly.

Example: Parallel processing. Suppose a program running on a 16-processor machine needs to multiply two large matrices together. For matrix multiply $C = A * B$, result entry $C_{(i,j)}$ is computed by taking the dot product of the i th row of A and the j th column of B : $C_{i,j} = \sum_{k=0}^{N-1} A_{(i,k)} B_{(k,j)}$. So, we can divide the work of computing C into 16 parts, and create one thread to compute each part, and then compute those parts on different processors in parallel. For example, thread 0 running on one processor might compute the upper left square:



3. **Coping with high latency I/O devices.** If a program has multiple threads, then when some are waiting for I/O to complete, others can be running.

User-level programs typically access I/O devices such as keyboards, screens, disks, and networks via system calls. If a system call blocks because it needs to wait for the I/O device, then the calling thread goes from the RUNNING state to the WAITING state, and the scheduler chooses another thread to run.

Example: Overlapping I/O and processing. For example, if a process running on a single-processor machine needs to encrypt 100 files, it might create two threads, each of which iterates through 50 files, reading one file, encrypting it, reading the next, and so on. Then, while one thread is waiting for a file to be read from disk, the other one can be using the processor to encrypt the file it read from disk.

Threads without kernel support

In the body of the text, we describe how an operating system can provide system calls so that a process can create multiple threads and have multiple schedulable PCBs in the kernel.

It is also possible to implement threads as a library completely at user level, without any operating system support.

The basic idea is simple. The threads library instantiates all of its data structures within the process: TCBs, the ready list, the finished list, and the waiting lists all are just data structures in the process’s address space. Then, calls to the threads library are just procedure calls (similar to how the same functions are implemented within a multi-threaded kernel as described above.)

What about interrupts and exceptions? Can we implement something like a timer interrupt so that we can suspend a long-running thread so that it does not monopolize the processor? Yes; on most operating systems, we can. Most operating systems implement *signal handlers* or something similar, which allows the operating system to trigger execution of specified handler code within a user-level process when a specified event like a timer interrupt or memory exception occurs.

So, first the user-level threads library for some process P uses a system call to register a signal handler with the kernel. When a hardware timer interrupt occurs, the hardware and kernel software save P 's register state and run the kernel's handler. Rather than restoring P 's register state and resuming P where it was interrupted, kernel's handler copies P 's saved registers into P 's address space (e.g., onto a signal stack that P has registered or onto P 's existing stack) and restore a slightly modified state with the stack pointer changed to point to where this state was saved and the instruction pointer changed to point to the signal handler code that P registered. Thus, P 's signal handler is invoked with the state of the previously running thread stored on its stack. It can then save that state to a TCB within the process and restore the state of some other TCB on the process's ready list.

This approach essentially virtualizes interrupts and exceptions, providing a user-level process with a very similar picture to the one the kernel gets when these events occur.

Why would you do this? Early libraries for threads often took this pure user-level approach for the simple reason that few operating systems supported multi-threaded processes. Even once operating system support for threads became widespread, pure user-level threads were sometimes used to minimize dependencies on the operating systems and to maximize portability; for example the earliest implementations of Sun's Java Virtual Machine (JVM) implemented what were called *green threads*, a pure user-level implementation of threads.

Why wouldn't you do this? From the kernel's point of view, a process that uses a library to implement pure user-level threads appears to be single-threaded no matter how many threads the user-level library creates.

As a result, pure user-level threads do not meet all of the goals of threads outlined at the start of this chapter. In particular, although pure user-level threads can be used for *program structure*, they do not help with *performance (parallel processing)* because the kernel only sees one "thread" that it can schedule. Similarly, they do not help with *performance (coping with high-latency I/O)* because if one thread blocks due to a long-running system call, the kernel does not have access to (and is unaware of) other TCBs that it could enable.

Today, most programs make use of kernel-supported threads rather than pure user-level threads. Major operating systems support threads and support them using fairly standard abstractions, so the past motivations are seldom compelling enough to compensate for the significant performance disadvantages.

Hybrid implementations. Although today few threads packages operate completely at user level, many split their work between the user-level library and the operating system kernel to reduce overheads by avoiding mode switches into the kernel for some of their operations.

Example: Hybrid thread join. As a simple example, rather than always having `thread_join()` make a system call to wait for the target thread to finish and return its exit value, we can have `thread_exit()` store its exit value in a data structure in the process's address space. Then, if the call to `thread_join()` happens after the targeted has exited, it can immediately return that stored value without having to make a system call and mode switch into the kernel. However, if the call to `thread_join()` precedes the call to `thread_exit()`, then the call to `thread_exit()` would make a system call to transition to the WAITING state and let some other thread run.

Example: Scheduler activations. A more sophisticated example is *scheduler activations*. Like the full user-level threads package described at the start of this sidebar, a thread system based on scheduler activations maintains its ready list and other data structures within the process and handles most thread management functions—`thread_create()`, `thread_destroy()`, `thread_yield()`, `thread_exit()`, and `thread_join()`, as well as the synchronization functions described in the next chapter—as procedure calls within the process.

Scheduler activations address the two problems noted above for pure user-level threads packages: they allow a process to run multiple threads in parallel on a multiprocessor, and they allow a process to hide I/O latency by running other threads when one thread blocks for I/O.

Scheduler activations solve these problems by allowing the kernel to activate the user-level scheduler when it wants to start running another one of the process's threads. Initially, the process does a system call to register its scheduler with the kernel; this is similar to registering a signal handler. Then, if the kernel is running on a multiprocessor machine and notices that a processor is available or if one of the process's threads blocks in a system call, the kernel can activate the process's scheduler in a way similar to how it would invoke a signal handler. The user-level scheduler activation code can then transition to running any of the process's READY threads.

Exercises

11. Using sthreads, write a program that creates several threads and that then determines whether the threads package on your system allocates a fixed size stack for each thread or whether each thread's stack starts at some small size and dynamically grows as needed.

Hints: You will probably want to write a recursive procedure that you can use to consume a large amount of stack memory. You may also want

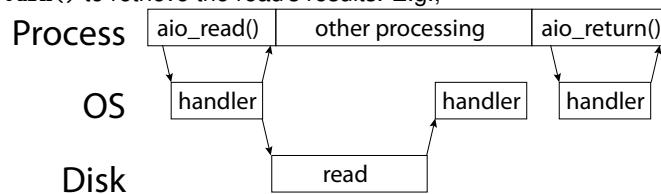
to examine the addresses of variables allocated to different threads' stacks. Finally, you may want to be able to determine how much memory has been allocated to your process; most operating systems have a command or utility that can show resource consumption by currently running processes (e.g., `top` in Linux, Activity Monitor in OSX, or Task Manager in Windows.)

4.5 Asynchronous I/O and event-driven programming

Although threads are a common way to express concurrency, they are not the only way. Asynchronous I/O and event-driven programming are one popular alternative. This approach allows a single-threaded program to cope with high-latency I/O devices by overlapping I/O with processing and other I/O.

The basic idea is to allow a process to make a system call to issue an I/O request but not wait for the result. At a later time the operating system provides the result to the process by calling a signal handler, by making a callback to code in the process, by placing the result in a queue in the process's memory, or by storing the result in kernel memory until the process makes another system call to retrieve it.

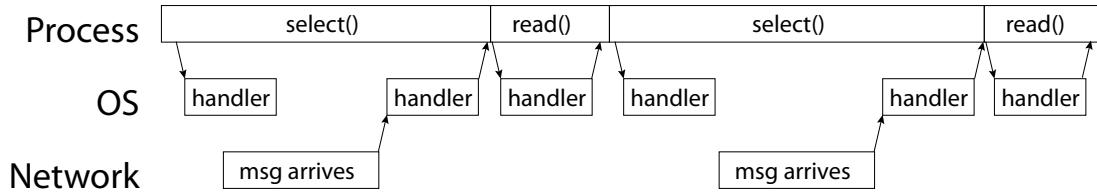
Example: Asynchronous disk read. Reading from disk can take tens of milliseconds, so in Linux, rather than issuing a `read()` system call that blocks until the requested blocks have been read from disk, a process can issue an `aio_read()` (asynchronous I/O read) system call, which tells the operating system to initiate the read from disk but which then immediately returns. Later, the process can call `aio_error()` to determine if the disk read has finished and `aio_return()` to retrieve the read's results. E.g.,



One common design pattern allows a single thread to interleave several different I/O-bound tasks by waiting for several different I/O events.

Example: Web server. Consider a web server with 10 active clients. Rather than create one thread per client and have each thread do a blocking `read()` on the network connection, an alternative is for the server to have one thread that does a `select()` call that blocks until *any* of the 10 network connections has data available to read; when the `select()` call returns, it provides the

thread with an identifier for a connection that has data available, and the thread can then `read()` from that connection, knowing that the read will immediately return. After processing the data, the thread then calls `select()` again to wait for the next message.



In servers, asynchronous I/O allows many concurrent tasks to be in progress at once. This gives rise to an *event-driven programming pattern* where a thread spins in a loop, and on each iteration it gets and processes the next I/O event. In order to be able to process each event, the thread typically maintains for each task a *continuation*, a data structure that keeps track of a task's current state and next step.

Definition: **event-driven programming pattern**

Definition: **continuation**

Example: Web server. Handling a web server request can involve a series of I/O steps: making a network connection, reading a request from the network connection, reading the requested data from disk, and writing the requested data to the network connection. If a single thread is handling requests from multiple different clients at once, it needs to keep track of the state of its processing for each request. For example, the network may divide a client's request into several packets so that the server needs to make several `read()` calls to assemble the full packet. The server may be doing this request assembly for multiple clients at once, so it needs to keep several per-client variables (e.g., a request buffer, the number of bytes expected, and the number of bytes received so far). When new data arrives, the thread uses the network connection's port number to identify which client sent the request and retrieves the appropriate client's variables using this port number/client ID. It can then process the data.

Event-driven programming v. threads. Although superficially different, overlapping I/O with computation and other I/O is fundamentally the same whether you use asynchronous I/O and event-driven programming or use synchronous I/O and threads. In either case, we block until the next I/O can be done, restore the state of the task that can make progress, execute the next step of that task, and save the task's state until it can take its next step. The differences are whether the state is stored in a continuation or thread control block and whether the state save/restore is done explicitly by the application or automatically by the threads system.

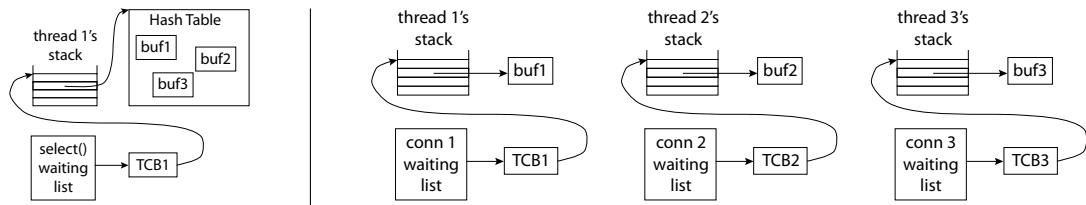
Example: Receiving from multiple connections. Consider a simple server that just collects incoming data from several clients. The pseudocode for the event-driven and thread-per-client cases are similar:

```
// Event-driven
Hashtable<Buffer*> *hash;
while(1){
    connection = use select() to find a
                readable connection ID
    buffer = hash.remove(connection);
    got = read(connection, tmpBuf, TMP_SIZE);
    buffer->append(tmpBuf, got);
    buffer = hash.put(connection, buffer);
}

// Thread-per-client
Buffer *b;
while(1){
    got = read(connection, tmpBuf, TMP_SIZE);
    buffer->append(tmpBuf, got);
}
```

When these programs execute, the system does essentially the same things. However, in the first case, it explicitly saves and restores each connection's state, while in the second case, the thread system saves and restores this state transparently.

The state in both cases is also essentially the same. In both cases, the application needs to keep one buffer per client connection. In the event-driven case, the application maintains a data structure (e.g., `hash` in the pseudocode) to keep track of the different clients' data structures. In the thread-per-client case, each thread has just one buffer to keep track of, and the operating system keeps track of the different threads' states.



To compare the event-driven and threads approaches, consider the three goals for threads discussed at the start of this chapter.

- **Performance: Coping with high-latency I/O devices.** Either approach—event-driven or threads—can be used to overlap I/O and processing. Which provides better performance?

In the past, the common wisdom has been that the event-driven approach could often be significantly faster for two reasons. First, the space and context switch overheads of the approach can be lower because a thread system must use generic code that allocates a stack for each thread's state and that saves and restores all registers on each context switch, but the event-driven approach allows programmers to allocate and save/restore just the state needed for each thread. Second, some past operating systems had inefficient or unscalable implementations of their thread systems, making it important not to create too many threads for each process.

Today, the comparison is less clear cut. Many systems now have large memories, so the cost of allocating a thread stack for each task is less

critical. For example, allocating 1000 threads with a 16KB stack for each thread on a machine with 4GB of memory would consume 0.4% of the machine’s memory. Also, most operating systems now have efficient and scalable threads libraries. For example, where the Linux 2.4 kernel had poor performance when processes had many threads, Linux 2.6 revamped the thread system, improving its scalability and absolute performance.

Anecdotal evidence suggests that the performance gap between the two approach has at least greatly narrowed and that for some applications the highly-optimized thread management code and synchronous I/O paths can beat the often less-optimized application code and asynchronous I/O paths. Performance has probably reached the point that for most application other factors (e.g., code simplicity and ease of maintenance) may be more important than raw performance. If performance is critical for a particular application, then—as is often the case—there is no substitute for careful benchmarking and measurements in making your decision.

- **Performance: Exploiting multiple processors.** The event-driven approach does not help a program exploit multiple processors. Note, however, that the event-driven and thread approaches can be combined: a program that wants to use n processors can have n threads, each of which uses the event-driven pattern to multiplex multiple I/O-bound tasks on each thread.
- **Program structure: Expressing logically concurrent tasks.** Whenever one compares two viable programming styles, one can find strong advocates for each approach. The situation is no different here, with some advocates of event-driven programming arguing that the synchronization required when threads share data makes threads more complex than events and advocates for threads arguing that threads provide a more natural way to express the control flow of a program than having to explicitly store a computation’s state in a continuation.

In our opinion, there remain cases where both styles are appropriate, and we use both in our own programs. That said, it seems to us that for most I/O-intensive programs, threads are preferable: they are often more natural, sufficiently efficient, and able to exploit multiple processors.

4.6 Conclusion and future directions

Concurrency is ubiquitous—many smartphones and the vast majority of servers, desktops, laptops, and tablets have 2 or more cores. Multi-threaded programming is a skill that every professional programmer needs to master.

Technology trends suggest that concurrent programming will only get more important over time. After several decades in which computer architects were

able to make individual processor cores run faster and faster, we have reached an era where individual cores are not getting much faster and where speedups will have to come from parallel processing.

Threads and event-driven programming are important models, but they are not the only ways to write parallel programs. Other approaches are also highly effective, particularly for certain classes of applications.

Definition: **Data parallel programming**

Definition: **SIMD (single instruction multiple data) programming**

- **Data parallel programs.** *Data parallel programming* or *SIMD (single instruction multiple data) programming* models allow a programmer to describe a computation that should be performed in parallel on many different pieces of data. Rather than having the programmer divide work among threads, the runtime system decides how to map the parallel work across the hardware's processors.

For example, to divide each element in an N-element array in half, you might write

```
forall(i in 0:N-1) array[i] = array[i]/2
```

and the runtime system would divide the array among processors to execute this computation in parallel.

Data parallel programming is frequently used in large data-analysis tasks. For example, the Hadoop system is widely-used, open source system that can process and analyze terabytes of data spread across hundreds or thousands of servers. SQL (Structured Query Language) is a standard language for accessing databases in which programmers specify the database query to perform and the database maps the query to lower-level operations and schedules those operations on its processors and disks.

Multimedia streams (e.g., audio, video, and graphics) often have large amounts of data on which similar operations are repeatedly performed, so data parallel programming is frequently used for media processing, and specialized hardware to support this type of parallel processing is common. Because they can be optimized for regular, data parallel programs, GPUs (Graphical Processing Units) can provide significantly higher rates of data processing. For example, in 2011 a Radeon HD5870 GPU is capable of 544 GFLOPS (billion (Giga) Floating Point Operations Per Second (double-precision)); for comparison, an Intel Core i7 980 XE CPU (a high end, general purpose processor) is capable of 109 double-precision GFLOPS.

Considerable research and development effort is currently going towards developing and using *General Purpose GPUs (GPGPUs)*, GPUs that have been extended to better support a wider-range of programs. It is still not clear what classes of programs can work well with GPGPUs and which require more traditional CPU architectures, but for those programs that can be ported to the more restrictive GPGPU programming model, the performance gains can be dramatic.

- **Distributed and parallel processing.** Many of the services we rely on today are implemented by large scale distributed systems. For example, when you use your browser to do a web search, hundreds or thousands of servers may work together to process your request and produce your search results, all in a few hundredths of a second.

Similarly, parallel scientific computations such as drug discovery, climate simulation, and geologic modeling can run on clusters of hundreds or thousands of machines, with threads on different machines coordinating their work by sending messages to each other over a high speed network.

There are entire courses devoted to parallel programming or and even to some of these individual techniques, and improving techniques for parallel programming is seen as an important research challenge. Over your career, all of these techniques will evolve and new ones are likely to appear.

Chapter 5

Synchronizing Access to Shared Objects

Multi-threaded programs extend the traditional, single-threaded programming model so that each thread provides a single sequential stream of execution composed of familiar instructions. If we only have *independent threads* that operate on completely separate subsets of state, then we can reason about each thread separately. In this case, writing and reasoning about independent threads differs little from writing and reasoning about a series of independent, single-threaded programs.

Definition: **independent threads**

However, most multi-threaded programs have both *per-thread* state (e.g., a thread's stack and registers) and *shared* state (e.g., shared variables on the heap). *Cooperating threads* read and write shared state.

Definition: **Cooperating threads**

Sharing state among threads is useful because it allows threads to communicate, to coordinate work, and to share information. For example, in our Earth Visualizer example in the previous chapter, once one thread has finished downloading a detailed image from the network, it needs to share that image data with a rendering thread that draws the new image on the screen.

Unfortunately, when cooperating threads use shared state, writing correct multi-threaded programs becomes much more difficult. Most programmers are used to thinking “sequentially” when reasoning about programs. For example, we often reason about the series of states traversed by a program as a sequence of instructions is executed. But this sequential model of reasoning breaks down in programs with cooperating threads for three reasons.

1. Program execution depends on the interleavings of threads' access to shared state.

For example, if two threads write a shared variable, one thread with the value

1 and the other with the value 2, the final value of the variable depends on which of the threads' writes finishes last.

Although this example is simple, the problem is severe because programs need to work for *any possible interleaving*. In particular, recall that thread programmers should not make any assumptions about the relative speed at which their threads operate.

Worse, as programs grow, there is a combinatorial explosion in the number of possible interleavings.

How can we reason about all possible interleavings of threads' actions in a multi-million line program.?

2. Program execution can be nondeterministic.

Different runs of the same program may produce different results. For example, the scheduler may make different scheduling decisions, the processor may run at a different frequency, or another concurrently running program may affect the cache hit rate. Even common debugging techniques—such as running a program under a debugger, recompiling with the `-g` option instead of `-O`, or adding a `printf()`—can change how a program behaves.

Jim Gray, the 1998 ACM Turing Award winner, coined the term *Heisenbugs* for bugs that disappear or change behavior when you try to examine them. Multi-threaded programming is a common source of Heisenbugs. In contrast *Bohrbugs* are deterministic and generally much easier to diagnose.

How can we debug programs whose behaviors change across runs?

3. Compilers and architectures reorder instructions.

Modern compilers and hardware will reorder instructions to improve performance. This reordering is generally invisible to single-threaded programs; compilers and processors take care to ensure that dependencies within a sequence of instruction are preserved. However, this reordering can become visible when multiple threads interact and observe intermediate states.

For example, consider the following code to compute `q` as a function of `p`.

Thread 1

```
p = someComputation();
pIsInitialized = true;
```

Thread 2

```
while(!pIsInitialized)
    ;
q = anotherComputation(p);
```

Although it might appear that this code ensures that `p` is initialized before `anotherComputation(p)` is called to compute `q`, it does not. To maximize instruction level parallelism, the hardware or compiler may set `pIsInitialized = true` before the computation to compute `p` has completed, and `anotherComputation(p)` may be computed using an unexpected value.

How can we reason about thread interleavings when compilers and hardware may reorder a thread's operations?

Structured synchronization. Given all of these challenges, multi-threaded code can introduce subtle, non-deterministic, and unreproducible bugs. This chapter describes a *structured synchronization* approach to sharing state in multi-threaded programs. Rather than scattering access to shared state throughout the program and attempting *ad hoc* reasoning about what happens when the threads' accesses are interleaved in various ways, we structure the program to facilitate reasoning about it and we use a set of standard synchronization primitives to control access to shared state. This approach gives up some freedom, but if you consistently follow the rules we describe, then reasoning about programs with shared state becomes much simpler.

The first part of this section elaborates on the challenges faced by multi-threaded programmers and on why it is futile to try to reason about all possible thread interleavings in the general, unstructured case.

- **Challenges.** Why are unstructured multi-threaded programs difficult to reason about?

The rest of the chapter describes how to structure shared objects in multi-threaded programs so that we can reason about them. We will describe aspects of this structure. First, we will structure a multi-threaded program's shared state as a set of *shared objects* that encapsulate the shared state and that define and limit how the state can be accessed. Second, to avoid *ad hoc* reasoning about the interleavings of access to the state variables within a shared object, we will describe how shared objects include a small set of *synchronization primitives* like locks and condition variables to coordinate access to their state by different threads. Third, to simplify reasoning about the code in shared objects, we will describe a set of *best practices* for writing the code that implements each shared object. Because the first two issues are so closely tied together, we address these two issues in two main sections:

- **Shared objects and synchronization variables.** How do we use synchronization variables like locks and condition variables to construct shared objects that encapsulate shared state?

- **Best practices.** Given the building blocks provided by synchronization variables, what is a systematic way to write and reason about the code for shared objects?

Finally, it is important to understand how the tools we use actually work, so we dive into the details of how synchronization primitives are implemented.

- **Implementing synchronization primitives.** How are locks and condition variables implemented?

Multi-threaded programming has a reputation for being difficult. We agree that it takes care. But, this chapter provides a set of simple rules that anyone can follow to implement objects that can be shared by multiple threads.

5.1 Challenges

The start of this section outlined the core challenge of multi-threaded programming: a multi-threaded program’s execution depends on the interleavings of different threads’ access to shared memory, which can make it difficult to reason about or debug these programs. In particular, cooperating threads’ execution may be affected by *race conditions*.

5.1.1 Race conditions

Definition: **race condition** A *race condition* is when the behavior of a program depends on the interleaving of operations of different threads. In effect, the threads run a race between their operations, and the results of the program execution depends on who wins the race.

Reasoning about even simple programs with race conditions can be difficult. To appreciate this, we will look at several extremely simple multi-threaded programs.

The world’s simplest cooperating-threads program. Suppose we run a program with two threads that do the following:

Thread 0	Thread 1
$x = 1;$	$x = 2;$

Q: What are the possible final values of x ?

A: $x = 1$ or $x = 2$ depending on which thread wins or loses the “race” to set x .

That was easy, so let’s try one that is a bit more interesting.

The world's second-simplest cooperating-threads program. Suppose that initially $y = 12$ and we run a program with two threads that do the following:

Thread 0	Thread 1
$x = y + 1;$	$y = y * 2;$

Q: What are the possible final values of x ?

A: We can get $x = 13$ if Thread 0 executes first or $x = 25$ if Thread 1 executes first. More precisely, we get $x = 13$ if Thread 0 reads y before Thread 1 updates y or we get $x = 25$ if Thread 1 updates y before Thread 0 reads y .

The world's third-simplest cooperating-threads program. Suppose that initially $x = 0$ and we run a program with two threads that do the following:

Thread 0	Thread 1
$x = x + 1;$	$x = x + 2;$

Q: What are the possible final values of x ?

A: Obviously, we can get $x = 3$. For example, Thread 0 can run to completion and then Thread 1 can start and run to completion. However, we can also get $x = 2$ or $x = 1$. In particular, when we write a single statement like $x = x + 1$, compilers on many processors produce multiple instructions such as (1) load memory location x into a register, (2) add 1 to that register, and (3) store the result to memory location x . If we mentally disassemble the above program into simple pseudo-assembly-code, we can see some of the possibilities.

One Interleaving	Another Interleaving	Yet Another Interleaving
<pre> load r1, x add r2, r1, 1 store x, r2 load r1, x add r2, r1, 2 store x, r2 final: x == 3 </pre>	<pre> load r1, x load r1, x add r2, r1, 1 add r2, r1, 2 store x, r2 store x, r2 final: x == 2 </pre>	<pre> load r1, x load r1, x add r2, r1, 1 add r2, r1, 2 store x, r2 store x, r2 final: x == 1 </pre>

Already, for this 2-line program, the complexity of reasoning about race conditions and interleavings is beginning to grow: not only would one have to reason about all possible interleavings of statements, but one would have to mentally disassemble the programs and reason about all possible interleavings of assembly instructions. (And if the compiler and hardware can reorder those instructions, things are even worse.)

5.1.2 Atomic operations

When we mentally disassembled the code in last example, we were able to reason about *atomic operations*, indivisible operations that cannot be interleaved or split with or by other operations.

Definition: **atomic operations**

On most modern architectures a load or store of a 32-bit word from or to memory is an atomic operation. So, our above analysis reasoned about interleaving of atomic loads and stores to memory.

Conversely, a load or store is not always an atomic operation. Depending on the implementation of the hardware, if two threads store the value of a 64-bit floating point register to a memory address, the final result might be the first value, the second value, or a mix of the two.

In the next subsection, we will give an example of reasoning about a program based on its atomic loads and stores to memory. Because of these challenges, we then move on to a better approach that raises the level of abstraction by constructing *shared objects* using *synchronization variables*.

5.1.3 Too much milk

Although one could, in principle, reason carefully about the possible interleavings of different threads' atomic loads and stores, doing so is tricky. To illustrate this, we will present three solutions to a simple problem called "Too much milk."

The too much milk problem models two roommates who share a refrigerator and who—as good roommates—make sure the refrigerator is always well stocked with milk. With such responsible roommates, the following scenario is possible:

Roommate 1's actions	Roommate 2's actions
3:00 Look in fridge; out of milk	
3:05 Leave for store	
3:10 Arrive at store	Look in fridge; out of milk
3:15 Buy milk	Leave for store
3:20 Arrive home; put milk away	Arrive at store
3:25	Buy milk
3:30	Arrive home; put milk away
3:35	Oh no!

We can model each roommate as a thread, and we can model the number of bottles of milk in the fridge with a variable in memory. The question is, if the only atomic operations on shared state are atomic loads and stores to memory,

Definition: **safety** can we devise a solution to the too much milk problem that ensures both *safety* (the program never enters a bad state) and *liveness* (the program eventually enters a good state.) Here, we strive for the following properties:

1. **Safety:** Never more than one person buys milk.

2. **Liveness:** If milk is needed, eventually somebody buys milk.

Simplifying assumption. Throughout our analysis in this section, we assume that the instructions are executed in exactly the order written—neither the compiler nor the architecture reorders instructions. Such an assumption is crucial for reasoning about the order of atomic load and store operations, but many modern compilers and architectures will violate it, so be careful about using this approach.

With modern machines and compilers, in addition to the challenges discussed in this section, one would also have to use *memory barriers* to constrain such reordering, further complicating the problem. We discuss memory barriers later.

Solution 1. We first present solution 1 of 3.¹

The basic idea is for a roommate to leave a note on the fridge before leaving for the store. The simplest way to leave this note—given our programming model that we have shared memory on which we can perform atomic loads and stores—is to set a flag when going to buy milk and to check this flag before going to buy milk. We might have each thread run the following code:

```
if(milk==0){           // if no milk
    if(note==0){       // if no note
        note = 1;      // leave note
        milk++;         // buy milk
        note = 0;        // remove note
    }
}
```

Unfortunately, this implementation can violate safety. For example, the first thread could execute everything up to and including the check of the `milk` value and then get context switched. Then the second thread could run through all of this code and buy milk. Finally, the first thread could be rescheduled, see that `note == 0` is true, leave the note, buy more milk, and remove the note, leaving the system with `milk == 2`.

```
if(milk==0){
    if(milk==0){
        if(note==0){
            note = 1;
            milk++;
            note = 0;
        }
    }
    if(note==0){
        note = 1;
        milk++;
        note = 0;
    }
}
```

Oh no!

¹Two more solutions are coming. The reader should be suspicious.

This “solution” makes the problem worse. The above code will usually work, but it may fail occasionally when the scheduler does just the right (or wrong) thing. We have created a Heisenbug that will cause the program to occasionally fail in ways that may be really hard to reproduce (e.g., probably only when the grader is looking at it or when the CEO is demonstrating the prototype to an important investor!).

Solution 2. We now present solution 2 of 3.²

In the above solution, we had to check the note before setting it, which led to the possibility of bad interleavings where one roommate had already made a decision to buy milk before notifying the other roommate of that decision. If we use two variables for the notes, a roommate can create a note before checking the other note and the milk and making a decision to buy. For example, we can do the following:

Path A	Path B
<pre>noteA = 1; // leave note if(noteB==0){ // if no note **A1** if(milk==0){// if no milk **A2** milk++; // buy milk **A3**} } noteA = 0; // remove note A</pre>	<pre>noteB = 1; // leave note if(noteA==0){ // if no note **B1** if(milk==0) { // if no milk **B2** milk++; // buy milk **B3**} } noteB = 0; // remove note B</pre>

If the first thread executes the Path A code and the second thread executes the Path B code, this protocol is safe; by having each thread write a note (“I might buy milk”) before deciding to buy milk, we ensure that we can never have both threads buy milk.

Although this intuition is solid, proving safety without enumerating all possible interleavings requires a bit of care.

Safety Proof. Assume for the sake of contradiction that the algorithm is *not* safe—both A and B buy milk. Consider the state of the two variables (`noteB`, `milk`) when thread A is at the line marked **A1** at the moment when the atomic load of `noteB` from shared memory to A’s register occurs. There are three cases to consider:

- Case 1: (1, *). Impossible because this state contradicts the assumption that thread A buys milk and reaches **A3**.
- Case 2: (0, > 0). Impossible because in this simple program the property `milk > 0` is a *stable property*—once it becomes true, it remains true forever. Thus, if `milk > 0` is true when A is at **A1**, A’s test at line **A2** will fail, and A will not buy milk, contradicting our assumption.

Definition: **stable property**

²The odds that this one will work also seem low, don’t they?

- Case 3: $(0, 0)$. We know that thread B must not currently be executing any of the lines marked **B1-B5**. We also know that either `noteA == 1` or `milk > 0` will be true from this time forward (`noteA OR milk` is also a stable property.) But, this means that B can not buy milk in the future (either the test at B1 or B2 must fail), which contradicts our assumption that both A and B buy milk. \square

Liveness. Unfortunately, solution 2 does not ensure liveness. In particular, it is possible for both threads to set their respective notes, for each thread to check the other thread's note, and for both threads to decide not to buy milk.

This brings us to solution 3.

Solution 3. We now present solution 3 of 3.³

Solution 2 was safe because a thread would avoid buying milk if there was any chance that the other thread *might* buy milk. For solution 3, we will make sure that at least one of the threads determines whether the other thread has bought milk or not before deciding whether or not to buy. In particular, we do the following:

Path A	Path B
<pre> noteA = 1; // leave note A while(noteB==1){ // wait for no note B ; // spin } if(milk==0){ // if no milk ***M** milk++; // buy milk } noteA = 0; // remove note A </pre>	<pre> noteB = 1; // leave note B if(noteA==0){ if(milk==0) { // if no milk milk++; // buy milk } } noteB = 0; // remove note B </pre>

We can show that solution 3 is safe using a similar argument to the one we used for solution 2.

To show that solution 3 is live, observe that code path B has no loops, so eventually thread B must finish executing the listed code and eventually, `noteB == 0` becomes true and remains true. Therefore, eventually thread A must reach line M and decide whether to buy milk. If it finds `M == 1`, then milk has been bought, and we are live. If it finds `M == 0`, then it will buy milk, and we are live.

5.1.4 Discussion

The above discussion shows that—assuming that instructions are executed in program order—it is possible to devise a solution to “Too much milk” that is both safe and live using nothing but atomic load and store operations on shared memory. However, the solution we developed is not terribly satisfying.

³This one will work, but even better answers come in the next section.

Memory barriers

Suppose you are writing low-level code that must reason about the ordering of memory operations. How can this be done on modern hardware and with modern compilers?

A *memory barrier* instruction prevents the compiler and hardware from reordering memory accesses across the barrier—no accesses before the barrier will be moved after the barrier and no accesses after the barrier will be moved before the barrier. One can add memory barriers to the “too much milk” solution or to Peterson’s algorithm to get code that works on modern machines with modern compilers. Of course, that may make such code even more complex.

Details of how to issue a memory barrier instruction depend on hardware and compiler details, but a good example is gcc’s `_sync_synchronize()` builtin, which tells the compiler not to reorder memory accesses across the barrier and to issue architecture-specific instructions that the underlying hardware will treat as a memory barrier.

- The solution is complex, and it requires careful reasoning to convince oneself that it works.
- The solution is asymmetric. Thread A executes slightly different code than Thread B. If we added more threads, more variations would be needed.

We should note that this limitation is not fundamental. For example, Peterson defined a symmetric solution to a more general version of “Too Much Milk” that works for any fixed number n of threads attempting to access a resource. More details on Petersen’s algorithm can be found elsewhere (e.g., http://en.wikipedia.org/wiki/Peterson%27s_algorithm.)

- The solution is inefficient. While thread A is waiting, it is “busy-waiting”—consuming CPU resources.
- The solution may fail if the compiler or hardware reorders instructions.

This limitation can be addressed through the use of memory barriers, but the need to add memory barriers further increases the complexity of trying to implement and reason about this type of algorithm; barriers also do not address the other limitations just mentioned. See the sidebar for a discussion of memory barriers.

5.1.5 A better solution

The next section will describe a better approach to writing programs in which multiple threads access shared state. We will write *shared objects* that use *synchronization objects* to coordinate different threads’ access to shared state.

Suppose, for example, we had a primitive called a *lock* that ensures that only one thread at a time can own a lock. Then, we can solve the too much milk problem by defining the class for a *Kitchen* object with the following method:

```
Kitchen::buyIfNeeded(){
    lock.acquire();
    if(milk == 0){      // if no milk
        milk++;         // buy milk
    }
    lock.release();
}
```

We will define locks and condition variables (another type of synchronization object) in the next section.

Exercises

1. Show that solution 3 to the Too Much Milk problem is safe—that it guarantees that at most one roommate buys milk.

5.2 Shared objects and synchronization variables

The above discussion should convince you that it is unappealing to try to write multi-threaded programs using just atomic loads and stores. Fortunately, decades of work have developed a much simpler approach that extends the

Threads Shared Objects

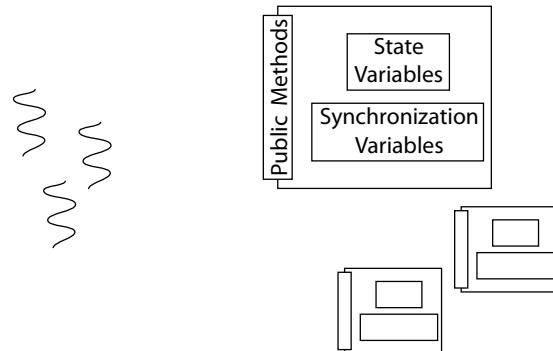


Figure 5.1: In a multi-threaded program, threads are separate from and operate on shared objects.

modularity of object oriented programming to multi-threaded programs: as Figure 5.1 illustrates, a multi-threaded program is build using *shared objects* and a set of threads that operate on them.

Definition: **Shared objects**

Shared objects are objects that can safely be accessed by multiple threads. All shared state in a program—including variables allocated on the heap (e.g., objects allocated with `malloc()` or `new()`) and static, global variables—should be encapsulated in one or more shared objects.

Shared objects extend traditional object-oriented programming in which objects hide their implementation details behind a clean interface. In the same way, shared objects hide the details of synchronizing the actions of multiple threads behind a clean interface. The threads using shared objects just need to understand the interface, they don't need to know how the shared object internally handles synchronization.

Just like regular objects, the shared objects are completely general—programmers implement shared objects for whatever modules, interfaces, and semantics their application needs. Each shared object's class defines a set of public methods on which threads operate. Then, to assemble the overall program out of these shared objects, each thread executes some “main loop” written in terms of actions on public methods of shared objects.

Note that since the shared objects encapsulate the program's shared state, the main-loop code that defines a thread's high-level actions doesn't concern itself with the details of synchronization. The programming model thus looks very much like it does for single-threaded code.

Implementing shared objects. Of course, internally the shared objects must handle the details of synchronization. As Figure 5.2 shows, shared objects are implemented in three layers.

- **Shared objects.** As in standard object-oriented programming, the shared objects define the application-specific logic and hide the internal implementation details. Externally, shared objects appear essentially the same as objects you define for single-threaded programs.
- **Synchronization variables.** Rather than trying to implement shared objects directly with carefully interleaved atomic loads and stores, shared objects include *synchronization variables* as member variables. Synchronization variables are stored in memory just like any other object, so they can be included in any data structure.

Definition:
**Synchronization
variables**

Synchronization variables are instances of carefully designed classes that provide broadly-useful primitives for synchronization. In particular, we build shared objects using two types of synchronization variables: *locks* and *condition variables*. We define these and describe how to construct them later.

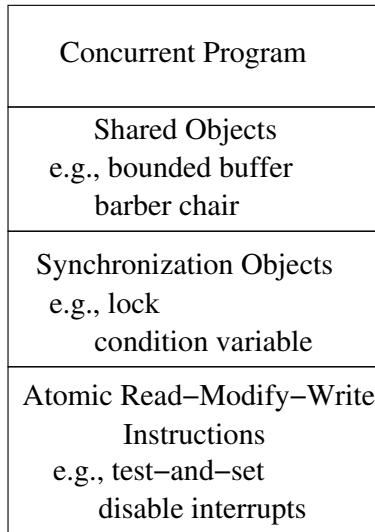


Figure 5.2: Multi-threaded programs are built with shared objects, shared objects are built using synchronization objects, and synchronization objects are implemented using atomic read-modify-write instructions.

Synchronization variables coordinate access to *state variables*, which are just the normal member variables of an object that you are familiar with from single-threaded programming (e.g., integers, strings, arrays, pointers, etc.)

Definition: **state variables**

Using synchronization variables simplifies implementing shared objects. In fact, not only do shared objects externally appear quite similar to traditional single-threaded objects, by implementing them with synchronization variables, we can also make their internal implementations quite similar to what you are used to from implementing single-threaded programs.

- **Atomic read-modify-write instructions.** Although the layers above benefit from a simpler programming model, it is not turtles all the way down. Internally, synchronization variables must manage the interleavings of different threads’ actions.

Rather than implementing synchronization variables such as locks and condition variables using only atomic loads and stores as we attempted to do for the Too Much Milk problem, modern implementations build synchronization variables using *atomic read-modify-write instructions*. An atomic read-modify-write instruction allows one thread to have exclusive access a memory location while the instruction’s read, modification, and write of that location are guaranteed not to be interleaved with any other thread’s access to that memory.

Definition: **atomic read-modify-write instructions**

Shared objects, monitors, and syntactic sugar

We focus on *shared objects* because object-oriented programming provides a good way to think about shared state: hide shared state behind public methods that provide a clean interface to threads and that handle the details of synchronization.

Although we use object-oriented terminology in our discussion, the ideas are equally applicable in non-object-oriented languages. For example, where a C++ program might define a class for shared objects that defines public methods to manipulate its private state variables and member variables in a well-defined way, a C program might define a struct that includes synchronization variables and state variables as fields, and—rather than having code that accesses the struct's fields scattered about—it might define a fixed set of functions that operate on the struct's fields.

Conversely, some programming languages build in even more support for shared objects than we describe here. When a programming language includes support for shared objects, a shared object is often called a *monitor*. Early languages with built-in support for monitors included Brinch Hansen's Concurrent Pascal and Xerox PARC's Mesa; today, Java includes built in monitor support via the `synchronized` keyword.

We regard the distinctions between procedural languages, object-oriented languages, and languages with built-in support for monitors as relatively unimportant syntactic sugar. We use the terms “shared objects” or “monitors” broadly to refer to a conceptual approach to constructing shared state that can and should be used regardless of the level of built in support in a particular programming language.

That said, for this book, our code and pseudo-code are based on C++'s syntax, which we believe provides the right level of detail for teaching the shared objects/monitors approach. We prefer teaching with C++ to, say, Java because we want to explicitly show where locks and condition variables are allocated and accessed rather than relying on operations hidden by a language's built in monitor syntax. Conversely, we prefer C++ to, say, C because we think C++'s support for object-oriented programming may help readers internalize the underlying philosophy of the shared object approach.

Scope and roadmap. As Figure 5.1 indicates, concurrent programs are built on top of shared objects. The rest of this chapter focuses on the three bottom layers of the figure—how to build shared objects using synchronization objects and how to build synchronization objects out of atomic read-modify-write instructions. Chapter ?? discusses issues that arise when composing multiple shared objects into a larger program.

5.3 Lock: Mutual Exclusion

Definition: lock A *lock* is a synchronization variable that provides *mutual exclusion*—when one thread holds a lock, no other thread can hold the lock (other threads are *excluded*.)

A program associates each lock with some subset of shared state and requires a thread to hold the lock when accessing that state. Then, only one thread can be accessing the shared state at a time.

Mutual exclusion greatly simplifies reasoning about programs because a thread can perform an arbitrary set of operations while holding a lock, and those operations *appear to be atomic* from the point of view of other threads. In particular, because a lock enforces mutual exclusion and because threads must hold the lock to access the shared state, no other thread will be able to observe an intermediate—they can only observe the state left after the lock has been released.

Example: Locking to group multiple operations. Consider, for example, a bank account object that includes a list of transactions and a total balance. To add a new transaction, we would acquire the account's lock, append the new transaction to the list, read the old balance, modify it, write the new balance, and release the lock. To query the balance and list of recent transactions, we would acquire the account's lock, read the recent transactions from the list, read the balance, and release the lock. Using locks this way would guarantee that one update or query completes before the next one starts so that a query always shows a balance that reflects the set of recent transactions shown.

It is much easier to reason about interleavings of atomic groups of operations rather than interleavings of individual operations for two reasons. First, there are (obviously) fewer interleavings to consider, so reasoning about interleavings on a coarser-grained basis reduces the sheer number of cases we have to worry about. Second, and more important, we can make each atomic group of operations correspond to the logical structure of the program, which allows us to reason about *invariants* not specific *interleavings*.

In particular, with shared objects we usually use one lock to guard all of an object's state, and we usually have each public method acquire the lock on entry and release the lock on exit. Then, reasoning about a shared class's code is similar to reasoning about a traditional class's code: we assume a set of invariants when a public method is called and reestablish those invariants before a public method returns. If we do a good job of defining our invariants, we can then reason about each method largely independently.

5.3.1 Lock: API and properties

A lock enables mutual exclusion by providing two methods: `Lock::acquire()` and `Lock::release()`. These operations are defined as follows:

- A lock can be in one of two states: BUSY or FREE.
- A lock is initially in the FREE state.

- `Lock::acquire()` waits until the lock is FREE , and then it atomically makes the lock BUSY.

Checking the state to see if it is FREE and setting the state to BUSY are together an *atomic operation*. Even if multiple threads are trying to acquire the lock, at most one thread will succeed—only one thread will observe that the lock is FREE and set it to BUSY; the other threads will just see that the lock is BUSY and wait.

- `Lock::release()` makes the lock FREE. If there are pending `acquire()` operations, then this state change causes one of them to proceed.

We will describe how to implement locks with the above properties in Section 5.5. But, assuming we can implement such a lock, solving the Too Much Milk problem is trivial. Both threads run the following code:

```
lock.acquire();
if(milk == 0){      // if no milk
    milk++;          // buy milk
}
lock.release();
```

Formal properties. The above definition describes the basic operation of a lock. A lock can be defined more precisely as follows.

We say that a thread *holds a lock* if it has returned from a lock's `acquire()` method more often than it has returned from a lock's `release()` method. We say that a thread *is attempting to acquire* a lock if it has called but not yet returned from a call to `acquire()` on the lock.

A lock should ensure the following three properties:

1. **Mutual Exclusion.** At most one thread holds the lock.
2. **Progress.** If no thread holds the lock and any thread attempts to acquire the lock, then eventually some thread succeeds in acquiring the lock.
3. **Bounded waiting.** If thread T attempts to acquire a lock, then there exists a bound on the number of times other threads successfully acquire the lock before T does.

The first property is a safety property. It says that we can use locks to enforce mutual exclusion on access to shared state.

The second and third properties are liveness properties. The second says that if a lock is FREE, *some* thread must be able to acquire it. The third defines a fairness property: any *particular* thread that wants to acquire the lock must eventually succeed in doing so. If the definitions above sound a bit stilted, it is because they are carefully crafted to avoid introducing subtle corner

```

// Attempt to insert an item. Return true on success.
// If the queue is full, return false.
bool
TSQueue::tryInsert(int item)
{
    bool ret = false;
    lock.Acquire();
    if(nFull < MAX){
        items[nextEmpty] = item;
        nFull++;
        nextEmpty = (nextEmpty + 1) % MAX;
        ret = true;
    }
    lock.Release();
    return ret;
}

// Attempt to remove an item. Return true on success.
// If the queue is empty, return false.
bool
TSQueue::tryRemove(int *item)
{
    bool ret = false;
    lock.Acquire();
    if(nFull > 0){
        *item = items[firstFull];
        nFull--;
        firstFull = (firstFull + 1) % MAX;
        ret = true;
    }
    lock.Release();
    return ret;
}

```

Code from TSQueue.h

Code from TSQueue.cc

Figure 5.3: The class definition for a simple shared object (a thread-safe queue.)

cases. For example, if a thread holding a lock never releases it, other threads can't make progress, so the *bounded waiting* condition is defined in terms of successful `acquire()` operations.

Non-property: Thread ordering. The *bounded waiting* fairness property defined above for locks is very weak. It guarantees that eventually a thread will get a chance to acquire the lock, but it does not, for example, promise that the threads will acquire the lock in FIFO order.

5.3.2 Locks and shared objects

As in standard object oriented programming, each shared object is an instance of a class that defines the class's state and the methods that operate on that state.

That state includes both state variables (e.g., ints, floats, strings, arrays, and

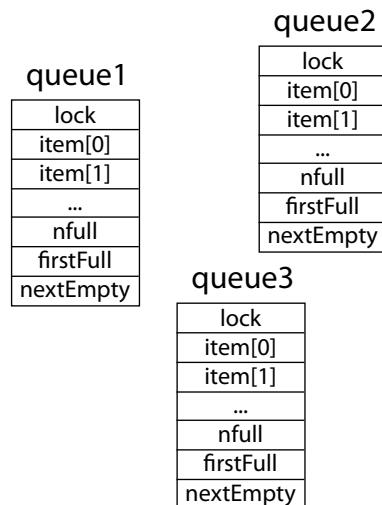


Figure 5.4: Three shared objects, each an instance of class `TSQueue`

pointers) and synchronization variables (e.g., locks). Thus, every time we use a class's constructor to produce another instance of a shared object, we allocate both a new lock and new instances of the state protected by that lock.

Figure 5.3 defines a simple shared object, a *thread-safe bounded queue*. Our implementation allows any number of threads to safely insert and remove items from it. As Figure 5.4 illustrates, a program can allocate multiple such queues (e.g., `queue1`, `queue2`, and `queue3`), each of which includes its own lock and state variables.

Notice that our queue stores only a bounded number of items. If the queue is full, an insert request returns an error flag. Similarly, if the queue is empty, a remove request returns an error flag. In the next section, we will show how *condition variables* would allow us to instead wait until the queue has room on insert or wait until an item is available on remove.

Also, to keep the example as simple as possible, we only allow items of type `int` to be stored in and removed from the queue, but the pattern can be used for queuing anything.

The `TSQueue` implementation defines a circular queue that stores data in an array, `items[]`. We maintain a number of simple invariants on the state. `nFull` indicates how many items are currently stored in the queue. If `nFull` is nonzero, then `firstFull` is the index of the oldest item in the queue. If `nFull` is less than `MAX`, then `nextEmpty` is the index where the next item to be inserted should be placed.

All of these variables are as they would be for a single-threaded version of this object. The `lock` allows `tryInsert()` and `tryRemove()` to atomically read

and write multiple variables just as a single-threaded version would.

Critical sections. A sequence of code that operates on shared state is a critical section. A *critical section* is a sequence of code that atomically accesses shared state. By ensuring that an object's lock is held while any of its critical sections is executed, we can ensure that each critical section on a collection of shared state appears to execute atomically.

Definition: **critical section**

Example: For the `TSQueue` class in Figure 5.3, the methods `tryInsert()` and `tryRemove()` each include a critical section that manipulates several of a `TSQueue` object's state variables (`items`, `nFull`, `firstFull`, and `nextEmpty`.)

Notice two things:

- Each class may define multiple methods that operate on the shared state defined by the class, so there may be *multiple critical sections per class*. However, for each instance of the class (i.e., for each object) only one thread holds the object's lock, *only one thread is actively executing any of the critical sections per shared object instance*.

Example: For the `TSQueue` class in Figure 5.3, if one thread calls `queue1.tryInsert()` and another thread calls `queue1.tryRemove()`, either the `insert()` will occur before the `remove()` or vice versa; these functions' accesses to object `queue1`'s state variables will not be interleaved.

- A program may create *multiple instances of a class*. Each instance is a shared object, and each shared object has its own lock. Thus, multiple threads may be active in the critical sections for different shared object instances.

Example: For the `TSQueue` class in Figure 5.3 and instances of that class in Figure 5.4, if one thread calls `queue1.tryInsert()`, another thread calls `queue2.tryRemove()`, and a third thread calls `queue3.tryInsert()`, all three threads may be simultaneously executing critical section code operating on *different instances* of the `TSQueue` class.

Using shared objects. Shared objects are allocated the same ways other objects—they can be dynamically allocated from the heap using `malloc()` and `new()` or they can be statically allocated in global memory by declaring static variables in the program.

Multiple threads need to be able to access shared objects. If shared objects are global variables, then a thread's code can just refer to an objects global name to reference it; the compiler computes the corresponding address. If shared objects are dynamically allocated, then each thread that uses an object needs a

```

int main(int argc, char **argv)
{
    TSQueue *queues[3];
    sthread_t workers[3];
    int ii, jj, ret;
    bool success;

    // Start the worker threads
    for(ii = 0; ii < 3; ii++){
        queues[ii] = new TSQueue();
        sthread_create_p(&workers[ii], putSome,
                         queues[ii]);
    }

    sthread_join(workers[0]);

    // Remove from the queues
    for(ii = 0; ii < 3; ii++){
        printf("Queue %d:\n", ii);
        for(jj = 0; jj < 20; jj++){
            success = queues[ii]->tryRemove(&ret);
            if(success){
                printf("Got %d\n", ret);
            }
            else{
                printf("Nothing there\n");
            }
        }
    }
}

```

```

void *putSome(void *tsqueuePtr)
{
    int ii;
    TSQueue *queue = (TSQueue *)tsqueuePtr;

    for(ii = 0; ii < 100; ii++){
        queue->tryInsert(ii);
    }
    return NULL;
}

```

Figure 5.5: This code creates three TSQueue objects and adds and removes some items from these queues. The full code is in TSQueueMain.cc. Note that rather than creating threads using the `sthread_create()` function introduced earlier, we use the `sthread_create_p()` variation so that we can pass a pointer to the newly created thread; in this case, we pass each newly created thread a pointer to the queue it will use.

pointer or reference to it. Two common ways to provide a pointer to a shared object to a thread are (1) providing a pointer to the shared object when the thread is created and (2) storing references to shared objects in other shared objects (e.g., containers). For example, a program might have a global, shared (and synchronized!) hash table that threads can use to store and retrieve references to other shared objects.

Example: Using the queue. Figure 5.5 shows a simple program that creates three queues and then creates some threads that insert into these queues. Finally, it removes 20 items out of each of the queues and prints the values it removes. The initial, main thread allocates the shared queues on the heap using `new()`, and each of the worker threads is provided a pointer to a shared queue when it is created.

Warning: Nothing prevents you from writing a program that allocates a shared object as an automatic variable in a procedure or method, but you should

not write programs that do this. Automatic variables (sometimes called “local variables” with good reason) are allocated on the stack as part of a procedure invocation. If one thread passes a pointer or reference to one of its automatic variables to another thread and later returns from the procedure where the automatic variable was allocated, then that second thread now has a pointer into a region of the first thread’s stack that may now be used for other purposes.

You might be tempted to argue that for a particular program, you know that the a procedure won’t return until all of the threads with which it is sharing an object are done using that object and that sharing one of the procedure’s local variables is safe. The problem with this argument is that someday the code may change, introducing a dangerous and subtle bug. When sharing dynamically-allocated variables, it is best to stay in the habit of only sharing variables from the heap and never sharing variables from the stack across threads.

Exercises

2. Precisely describe the set of possible outputs that could occur when the program shown in Figure 5.5 is run.
3. Suppose that a programmer mistakenly creates an automatic (aka local) variable v in one thread t_1 and passes it to another thread t_2 . Is it possible for a write by t_1 to some variable other than v will change the state of v as observed by t_2 ? If so, explain how this can happen and give an example. If not, explain why not.
4. Suppose that a programmer mistakenly creates an automatic (aka local) variable v in one thread t_1 and passes it to another thread t_2 . Is it possible for a write by t_2 to v will cause t_2 to execute the wrong code? If so, so, explain how. If not, explain why not.

5.4 Condition variables: Waiting for a change

Condition variables provide a way for one thread to wait for another thread to take some action. For example, in the thread safe queue example in Figure 5.3, rather than returning an error when we try to remove an item from an empty queue, we might want to wait until the queue is non-empty and always return an item.

Similarly, a web server might want to wait until a new request arrives, a word processor might want to wait for a key to be pressed, a weather simulator’s coordinator thread might wait for the worker threads calculating temperatures

```
// Naive, polling-based implementation of
// remove that waits so it can always return
// an item.
int
TSQueue::remove()
{
    int ret;
    bool empty;
    do{
        empty = tryRemove(&ret);
    } until(!empty);
    return ret;
}
```

Figure 5.6: A naive, polling-based implementation of `remove()` from a bounded queue that retries in a loop until it succeeds in removing an item. The method `tryRemove()` is defined in Figure 5.3.

in each region to finish, or in our Earth Visualizer example, a thread in charge of rendering part of the screen might wait either for a user input that shifts the scene or for new data to update the view.

Broadly speaking, what we want to do in all of these cases is have a thread wait for something else to change the state of the system so that it can make progress.

One way to have a thread wait for another thread to act would be to *poll*. Definition: **poll**—repeatedly check the state of interest. For example, `TSQueue` could wrap `tryRemove()` in a polling loop to provide a `remove()` method that always returns an item as shown in Figure 5.6. Unfortunately, such an approach can be inefficient because waiting threads continually loop, consuming CPU cycles without making useful progress. Worse, it might delay the scheduling of the other threads—perhaps exactly the ones for which the looping threads are waiting.

A “fix” to the polling-based approach is to add a delay. For example, in Figure 5.6, we might sleep, yielding the processor, for 100ms after each unsuccessful `tryRemove()` call so that there is a 100ms delay after a failed attempt.

This approach has two problems. First, although it reduces the inefficiency of polling, it does not eliminate it. Suspending and scheduling a thread imposes nontrivial overheads, and if a program has a large collection of threads polling every few tens or hundreds of milliseconds, they may still consume significant resources. Second, periodic polling adds latency. In our hypothetical Google Earth example, if the thread waiting for keyboard input waited 100ms between each check, the application might become noticeably more sluggish.

As an extreme example, one of the authors once had an employee implement a network server that provided several layers of processing, where each layer had a thread that received work from the layer above and sent the work to the layer below. Measurements of the server showed surprisingly bad performance; we

```

SharedObject::someMethodThatWaits()
{
    lock.acquire();
    // ... read and/or write shared state here ...
    while(!testOnSharedState()){
        cv.wait(&lock);
    }
    assert(testOnSharedState());
    // ... read and/or write shared state here ...
    lock.release();
}

```

Figure 5.7: Design pattern for a method that `wait()`'s using a condition variable.

expected each request to take a few milliseconds but instead each request was taking half a second. Fortunately, the performance was so horrible that it was easy to track down the problem: layers were passing work to each other through queues much like the `TSQueue` shown above, and `remove()` was implemented as a polling loop with a 100ms delay. With five such layers of processing, the server became unusably slow. Fortunately, the fix was simple: use condition variables, which we will now define.

5.4.1 Condition variable definition

A *condition variable* is a synchronization object that enables a thread to efficiently wait for a change to shared state that is protected by a lock. A condition variable has three methods:

Definition: **condition variable**

- `CV::wait(Lock *lock)` atomically *releases the lock* and *suspends execution of the calling thread*, placing the calling thread on the condition variable's *waiting queue*. Later, when the calling thread is reenabled, it *reacquires the lock* before returning from the `wait()` call.
- `CV::signal()` takes one waiting thread off the condition variable's *waiting queue* and marks it as eligible to run (i.e., it puts the thread on the scheduler's ready list.)
- `CV::broadcast()` takes all waiting threads off the condition variable's *waiting queue* and marks them as eligible to run.

Notice that a condition variable is always associated with a lock. One uses a condition variable to wait for a change to shared state, and updates to shared state are protected with a lock. Thus, the condition variable API is carefully designed to work in concert with mutual exclusion locks.

In particular, the standard design pattern is for a shared object to include a lock and zero or more condition variables. Then, a method that waits using

```

SharedObject::someMethodThatSignals()
{
    lock.acquire();
    // ... read and/or write shared state here ...

    // If state has changed in a way that could
    // allow another thread to make progress,
    // signal (or broadcast).
    cv.signal();

    lock.release();
}

```

Figure 5.8: Design pattern for a method that `signal()`'s using a condition variable. The pattern for `broadcast()` is similar.

a condition variable will work as shown in Figure 5.7. In this code, the calling thread first acquires the lock and then can read and write the shared object's state variables. To wait until `testOnSharedState()` passes, the thread calls `wait()` on the shared object's condition variable `cv`. Later, once the calling thread runs again and sees `testOnSharedState()` pass, it can do whatever it is it wants to do, release the lock, and return.

Figure 5.8 shows complementary code that changes the shared object's state in a way that might allow a waiting thread to make progress and then signals that thread with the condition variable.

Condition variables integrate with locks. Notice that a waiting thread is always waiting for the state of a shared object to change, so it must inspect the object's state in a loop. So, the condition variable's `wait()` method releases the lock (to let other threads change the state of interest) and then reacquires the lock (to check that state again.)

Similarly, the only reason for a thread to `signal()` (or `broadcast()`) is that it just changed the state in a way that may be of interest to a waiting thread. In order to make such a change to shared state, the thread must hold the lock, so `signal()` and `broadcast()` are always called while holding a lock on the state that was just changed.

Discussion. As just indicated, condition variables have been carefully designed to work in tandem with mutual exclusion locks and shared state. The precise definition of condition variables therefore includes three properties worth additional comment:

1. A condition variable is *memoryless*; the condition variable, itself, has no internal state other than a queue of waiting threads.

Condition variables do not need state of their own because they are always used within shared objects that define their own state. Condition variables

therefore provide a way to wait for changes to the enclosing object's state, but they do not have interesting state of their own.

If no threads are currently on the condition variable's `waiting queue`, a `signal()` or `broadcast()` call has no effect. In particular, the condition variable does not "remember" an earlier call to `signal()` or `broadcast()`, and a later call to `wait()` will still block until `signal()` or `broadcast()` is called again.

2. *Wait() atomically releases the lock.*

A thread always calls `wait()` while holding a lock, and the call to wait *atomically* releases the lock and puts the thread on the condition variable's `waiting queue`. The atomicity ensures that there is no separation between checking the shared object's state, deciding to wait, adding the waiting thread to the condition variable's queue, and releasing the lock so that the other thread can access the shared object to change its state and signal.

Conversely, if threads released the lock before calling `wait()`, we would have to contend with the possibility that a thread misses a signal or broadcast and waits forever.

Example: A missed signal.. For example, consider the case where thread T_1 checks an object's state and decides to wait, so it releases the lock in anticipation of calling `cv.wait()` on the shared object's condition variable `cv`. Then thread T_2 changes the object's state to what T_1 wants and calls `cv.signal()`, but `cv` has no waiting threads so the call to `signal()` has no effect. Finally T_1 calls `cv.wait()`, has its execution suspended, and is put on `cv`'s list of waiting threads. Unfortunately, the lack of atomicity means that T_1 missed the signal and is now waiting even though T_2 has already changed the state to what T_1 wants.

Because `wait()` releases the lock, any invariants that should be true for the object must be reestablished before calling `wait()`. Similarly, code must not assume more than the normal invariants that are always true for an object when `wait()` returns; other than that, just because something was true before `wait()` was called does not mean it remains true when `wait()` returns—because `wait()` releases the lock, other threads may access the object and change its state during a call to `wait()`. In practice most methods that `wait()` are simple enough that they follow these restrictions naturally, but be aware of the potential pitfall.

3. When a waiting thread is reenabled via `signal()` or `broadcast()`, it may not run immediately.

When a waiting thread is re-enabled, it is moved to the scheduler's ready queue with no special priority, and the scheduler may run it at some later time. Furthermore, when the thread finally does run, it must reacquire the lock, which means that other threads may have acquired and released the lock in the meantime. Therefore, even if the desired predicate was true

when `signal()` or `broadcast()` was called, it may no longer be true when `wait()` returns. This may seem like a small window of vulnerability, but we need to design concurrent programs to work with all possible schedules. Otherwise, programs may fail sometimes, but not always, making debugging very difficult. See the sidebar on *Hansen v. Hoare semantics* on page 206 for a discussion of the history behind this property.

Note to programmers. The points above have an important implication for programmers: `wait()` *must always be called from within a loop*.

Because `wait()` releases the lock and because there is no guarantee of atomicity between `signal()/broadcast()` and the return of a call to `wait()`, when a thread returns from `wait()`, there is no guarantee that the checked-for state holds. Therefore, a waiting thread must always wait in a loop, rechecking the state until the desired predicate holds. Thus, the design pattern is

```
...
while(predicateOnStateVariables(...)){
    wait(&lock);
}
...
```

and not

```
...
if(predicateOnStateVariables(...)){
    wait(&lock);
}
...
```

There are fundamentally two reasons why condition variables are designed to impose this requirement: *simplifying implementation* and *improving modularity*.

Implementation: As noted above, when a waiting thread is reenabled, it may not run immediately, other threads may access the shared state before it runs, and the desired predicate on the shared state may no longer hold when `wait()` finally does return.

This definition simplifies the implementation of condition variables without hurting the complexity of the code that uses them. No special code is needed for scheduling—we just put a signalled thread onto the scheduler’s ready list and let it run whenever the scheduler chooses it. Similarly, no special code is needed for acquiring the lock—we just have the thread call `Lock::acquire()` when it is rescheduled; just as with any attempt to acquire a lock, it may succeed immediately, or it may have to wait while other threads acquire and release the lock.

Some implementations go even further and warn that a call to `wait()` may return even if no thread has called `signal()` or `broadcast()`. So, not only is it possible that the desired predicate on the state *is no longer true*, it is possible that the desired predicate on the state *was never true*. For example, Sun’s Java JDK1.5’s interface to condition variables allows for “spurious wakeups”:

When waiting upon a Condition, a “spurious wakeup” is permitted to occur, in general, as a concession to the underlying platform semantics. This has little practical impact on most application programs as a Condition should always be waited upon in a loop, testing the state predicate that is being waited for. An implementation is free to remove the possibility of spurious wakeups but it is recommended that applications programmers always assume that they can occur and so always wait in a loop.

(From http://download.oracle.com/docs/cd/E17476_01/javase/1.5.0/docs/api/index.html)

Modularity: Waiting in a loop that checks the shared state makes shared objects’ code more modular because we can reason about when the thread will continue by looking just at the `wait()` loop; in particular, we do not need to examine the rest of the shared object’s code to understand where and why calls to `signal()` and `broadcast()` are done to know the postcondition for `wait()` loop. For example, in Figure 5.7, we know the `assert()` call will never fail without having to look at any other code.

Not only does always waiting in a loop simplify writing and reasoning about the code that waits, it simplifies writing and reasoning about the code that signals or broadcasts because you never have to worry that signaling at the wrong time will cause a waiting thread to proceed when it shouldn’t. `Signal()` and `broadcast()` can be regarded as *hints* that it *might* be a good time to try to proceed, but if the hints turn out to be wrong, no damage is done. You can always convert a `signal()` to a `broadcast()` add any number of `signal()` or `broadcast()` calls without changing the semantics of the object. Avoiding extra `signal()` and `broadcast()` calls may matter for performance, but not for correctness.

Bottom line: Given the range of implementations that are possible and given the modularity benefits, `wait()` must always be done from within a loop that tests the desired predicate.

5.4.2 Thread life cycle revisited

In Chapter 4 on page 152, we discussed how a thread can switch between the READY, WAITING, and RUNNING states. We can now explain the WAITING state in more detail.

A RUNNING thread that calls `Cond::wait()` is put in the WAITING state. This is typically implemented by moving the thread’s thread control block (TCB) from the scheduler’s ready queue to a queue of waiting threads associated with that condition variable. Later, when some other RUNNING thread calls `Cond::signal()` or `Cond::broadcast()` on that condition variable, one (if

Hansen v. Hoare semantics

In modern condition variables, `signal()` or `broadcast()` calls take waiting threads from a condition variable's waiting queue and put them on the scheduler's ready list. Later, when these threads are scheduled, they may block for some time while they try to reacquire the lock. Thus, modern condition variables implement what are sometimes called *Hansen semantics* (for Brinch Hansen, who first defined such condition variables) or *Mesa Semantics* (for Mesa, an early programming language at Xerox PARC that implemented these semantics.)

C.A.R. "Tony" Hoare proposed a different definition for condition variables. Under *Hoare semantics*, when a thread calls `signal()`, execution of the signaling thread is suspended, the ownership of the lock is immediately transferred to one of the waiting threads, and execution of that thread is immediately resumed. Later, when that resumed thread releases the lock, ownership of the lock reverts to the signaling thread, whose execution continues.

Thus, under Hoare semantics, signaling is atomic with the resumption of a waiting thread, and a signaled thread may assume that the state has not changed since the signal that woke it up was issued. So, where under Hansen semantics, waiting is always done in a loop (e.g., `while(predicate()){cv.wait();}`), under Hoare semantics, waiting can be done with a simple conditional (e.g., `if(predicate()){cv.wait();}`)

There are debates about which semantics are better. Some argue that the atomicity of signaling and resuming a waiting process makes it easier to prove liveness properties of programs under Hoare semantics. For example, if I know that one thread is waiting on a condition, and I do a signal, I know that waiting thread will resume and make progress (and not some other late-arriving thread.)

For what it is worth, the authors of this book come down strongly on the side of Hansen supporters. The modularity advantages of Hansen semantics greatly simplify reasoning about an object's core safety properties. For the properties we care most about (i.e., the safety properties that the threads only proceed when they are supposed to) and for large programs where modularity matters, Hansen semantics seem vastly preferable to us. Furthermore, in cases where liveness is a concern, you can implement explicit queuing to manage the order that waiting threads access an object (we leave this as an exercise for the reader.)

Regardless of which semantics are better, as a practical matter the debate has been settled: essentially all systems use Hansen semantics, and we know of no widely-used system that implements Hoare semantics. This resolution of the debate may be due as much to a desire for simpler implementations than to a universal consensus regarding the "right" semantics. Furthermore, programmers that program assuming the weaker Hansen semantics (e.g., always writing `while(predicate()) cv.wait(&lock);`) will write programs that will work under either definition. (And note that the overhead from the "extra" check of the predicate upon return from `wait()` in a `while` loop rather than an `if` conditional is unlikely to be significant compared to the signaling and scheduling overheads being paid in any event.) In any event, as a programmer, you won't go wrong if you write your code assuming Hansen semantics.

`signal()` or all (if `broadcast()`) of the TCBs on that condition variable's waiting queue are moved to the scheduler's ready list. This changes those threads from the WAITING state to the READY state. At some later time, the scheduler selects a READY thread and runs it by moving it to the RUNNING state.

Locks are similar. `Lock::acquire()` on a busy lock puts caller into the WAITING state, with the caller's TCB on a list of waiting TCBs associated with the lock. Sometime later, when the lock owner calls `Lock::release()`, the TCB is moved to the ready list and the thread transitions to the READY state.

Notice that threads that are RUNNING or READY have their state located at a pre-defined, “global” location like the CPU (for a RUNNING thread) or the scheduler's list of ready threads (for a READY thread). However, threads that are WAITING typically have their state located on some per-lock or per-condition-variable queue of waiting threads. Then, a `Cond::signal()`, `Cond::broadcast()`, or `Lock::release()` call can easily find and reenable a waiting thread for that particular condition variable or lock.

5.4.3 Example: Blocking bounded queue

As an example, consider the bounded queue described in Section 5.3.2. Suppose we change the interface so that `insert()` blocks until there is room to insert an item and `remove()` blocks until there is an item to remove. `BBQ.h` in Figure 5.9 defines the *blocking bounded queue*'s interface and public methods. `BBQ.cc` in Figure 5.10 defines the queue's implementation.

Notice that as in `TSQueue`, we acquire and release the lock at the beginning and end of the public methods (e.g., `insert()` and `remove()`). Now, however, we can (atomically) release the lock and wait if there is no room in `insert()` or there is no item in `remove()`. Before returning, `insert()` signals on `itemAdded` since a thread waiting in `remove()` may now be able to proceed; similarly, `remove()` signals on `itemRemoved` before it returns.

We `signal()` rather than `broadcast()` because each `insert()` allows at most one `remove()` to proceed and vice versa.

```
#include "Cond.h"

const int MAX = 10;

class BBQ{
private:
    // Synchronization variables
    Lock lock;
    Cond itemAdded;
    Cond itemRemoved;

    // State variables
    int items[MAX];
    int nFull;
    int firstFull;
    int nextEmpty;

public:
    BBQ();
    ~BBQ() {};
    void insert(int item);
    int remove();

private:
    // Private methods are called with lock already held
    inline bool isFull() {return (nFull == MAX) ? true : false;};
}
```

Figure 5.9: `BBQ.h` defines the interface and member variables for our blocking bounded queue.

```

/*
 * BBQ.cc -- Blocking Bounded Queue.
 */
#include <assert.h>
#include <pthread.h>
#include "BBQ.h"

BBQ::BBQ()
{
    nFull = 0;
    firstFull = 0;
    nextEmpty = 0;
}

// Wait until there is room
// then insert an item.
void
BBQ::insert(int item)
{
    lock.Acquire();

    while(isFull()){
        itemRemoved.Wait(&lock);
    }
    assert(!isFull());
    items[nextEmpty] = item;
    nFull++;
    nextEmpty = (nextEmpty + 1) % MAX;

    itemAdded.Signal(&lock);
    lock.Release();
    return;
}

// Wait until there is an item
// then remove an item.
int
BBQ::remove(void)
{
    int ret;
    lock.Acquire();
    while(isEmpty()){
        itemAdded.Wait(&lock);
    }
    assert(!isEmpty());
    ret = items[firstFull];
    nFull--;
    firstFull = (firstFull + 1) % MAX;

    itemRemoved.Signal(&lock);
    lock.Release();
    return ret;
}

```

Figure 5.10: BBQ.cc defines the implementation of our blocking bounded queue.

Semaphores considered harmful

During system conception it transpired that we used the semaphores in two completely different ways. The difference is so marked that, looking back, one wonders whether it was really fair to present the two ways as uses of the very same primitives. On the one hand, we have the semaphores used for mutual exclusion, on the other hand, the private semaphores.

(From Dijkstra "The structure of the 'THE'-Multiprogramming System" *Communications of the ACM* v. 11 n. 5 May 1968.)

In this book we focus on constructing shared objects using locks and condition variables for synchronization. Another widely used type of synchronization variable is a *semaphore*.

Semaphores were introduced by Dijkstra to provide synchronization in the THE operating system, which (among other advances) explored structured ways of using concurrency in operating system design.

Semaphores are defined as follows:

- A semaphore has a non-negative value
- When a semaphore is created, value can be set to any non-negative integer
- `Semaphore::P()` waits until value is positive. When value is positive, it atomically decrements value by 1.
- `Semaphore::V()` increments the value by 1, and if any threads are waiting in `P()`, one such thread is atomically enabled with its call to `P()` succeeding in decrementing the value and returning.

Note that `P()` is an atomic operation—the read that observes the positive value is atomic with the update that decrements it. Also note that if `V()` enables a thread in `P()`, then `P()`'s increment and `V()`'s decrement of value are atomic—no other thread can observe the incremented value, and the thread in `P()` is guaranteed to decrement the value and return.

The names `P()` and `V()` are historical artifacts. Sorry.

Given this definition, semaphores can be used for either mutual exclusion (like locks) or waiting for another thread to do something (like condition variables.)

To use a semaphore like a mutual exclusion lock, initialize it to 1. Then `Semaphore::P()` is equivalent to `Lock::acquire()` and `Semaphore::V()` is equivalent to `Lock::release()`.

Using a semaphore for more general waiting is trickier. Normally (but not always), you initialize the semaphore to 0. Then `Semaphore::V()` is *similar to* `Cond::wait(&lock)` and `Semaphore::P()` is *similar to* `Cond::signal()`. However, there are important differences. First, `Cond::Wait(&lock)` atomically releases a lock, so you can check a shared object's state while holding the lock and then atomically suspend execution; conversely `Semaphore::V()` does not release an associated mutual exclusion lock, so you have to carefully construct the program so that you can release the lock and then call `V()` without allowing intervening operations to cause confusion. Second, whereas a condition variable is stateless, a semaphore has a value, so if no threads are waiting, a call to `Cond::signal()` has no effect, while a call to `Semaphore::P()` increments the value, which will cause the next call to `Semaphore::V()` to proceed without blocking.

Semaphores considered harmful.⁴ Our view is that programming with locks and condition variables is superior to programming with semaphores, and we advise you to always write your code using those synchronization variables for two reasons.

First, using separate lock and condition variable classes makes code more self-documenting and easier to read. As the quote from Dijkstra above notes, there really are two abstractions here, and code is clearer when the role of each synchronization variable is made clear through explicit typing.

Second, a stateless condition variable bound to a lock turns out to be a better abstraction for generalized waiting than a semaphore. By binding a condition variable to a lock, we can conveniently wait on any arbitrary predicate on an object's state. In contrast, semaphores rely on carefully mapping the object's state to the semaphore's value so that a decision to wait or proceed in `P()` can be made entirely based on the value, without holding a lock or examining the rest of the shared object's state.

Although we do not recommend writing new code with semaphores, code based on semaphores are not uncommon, especially in operating systems. So, it is important to understand the semantics of semaphores and be able to read and understand semaphore-based code written by others.

One exception. There is one situation in which semaphores are often superior to condition variables and locks: synchronizing communication between an I/O device and threads running on the processor. In this situation, there is often a data structure shared by the hardware device and the operating system software, and it is often not possible to require the hardware to acquire a lock on that data structure before updating it. Instead, the data structure, hardware, and device drivers are designed to work with carefully-ordered atomic memory operations.

If a hardware device needs attention, it updates the shared data structure and then may need to cause some waiting operating system thread to run.

To trigger this waiting thread, one might consider using a condition variable and calling `Signal()` without holding the lock (this is sometimes called a *naked notify*.) Unfortunately, there is a corner case: suppose that the operating system thread first checks the data structure, sees that no work is currently needed, and is just about to call `Wait()` on the condition variable. At that moment, the hardware updates the data structure with the new work and calls `Signal()`. Thus, when the thread calls `Wait()`, the signal has already occurred and the thread waits—possibly for a long time.

The solution is for devices to enable operating system threads using semaphores instead. Because semaphores are stateful, a `Signal()` (or `V()`) cannot be lost. A common approach is for the hardware to send information to the software thread via a bounded queue that is synchronized using semaphores.

⁴Edsger Dijkstra is the author of the short note “Go To Statement Considered Harmful”, *Communications of the ACM*, v. 11 n. 3, March 1968, pp 147–148.

Exercises

5. Assuming Hansen semantics for condition variables, our implementation of the blocking bounded queue in Figure 5.10 does not guarantee freedom from starvation: if a continuous stream of threads makes `insert()` (or `remove()`) calls, it is possible for a waiting thread to wait forever. For example, a thread may call `insert()` and wait in the `while(isFull())` loop; then, every time another thread calls `remove()` and signals on the `itemRemoved` condition variable, a *different* thread might call `insert()`, see that the queue is not full, and insert an item before the waiting thread resumes. Then, when the waiting thread resumes, it will retest the `isFull()` predicate, see that the queue is full, and `wait()`.

Prove that under Hoare semantics and assuming that when a signal occurs, it is the longest-waiting thread that is resumed, our implementation of BBQ ensures freedom from starvation. That is, that if a thread waits in `insert()`, then it is guaranteed to proceed after a bounded number of `remove()` calls complete, and vice versa.

6. As noted in the previous problem, our implementation of the blocking bounded queue in Figure 5.10 does not guarantee freedom from starvation. Modify the code to ensure freedom from starvation so that if a thread waits in `insert()`, then it is guaranteed to proceed after a bounded number of `remove()` calls complete, and vice versa. **Note:** Your implementation must work under *Hansen/Mesa semantics* for condition variables.

5.5 Implementing synchronization objects

We have described two types of synchronization variables, locks and condition variables. Given these synchronization variables, we can implement shared objects to coordinate access to shared state. In this section, we describe how to implement these important building blocks.

Recall from Chapter 4 that threads can be implemented for the kernel or for user-level processes. We will initially describe how to implement locks for in-kernel threads; at the end of this section we discuss the changes needed to support these abstractions for threads in user-level processes.

5.5.1 Implementing locks

Recall from Section 5.3 that a lock allows us to group an arbitrary sequence of operations on shared state into an atomic unit by calling `Lock::acquire()`

at the start of the sequence and `Lock::release()` at the end. To enable this abstraction, each lock has a state (FREE or BUSY) and a queue of zero or more threads waiting for the lock to become FREE.

The discussion of Too Much Milk showed that it is both complex and costly to implement locks with just memory reads and writes, so modern implementations use more powerful hardware primitives that allow us to atomically read, modify, and write pieces of the lock's state. We will look at locks based on two primitives: *disabling interrupts*, which can make a sequence of instructions atomic with respect to a single processor, and *atomic read-modify-write* instructions, which allow a single instruction to atomically read and update a word of memory with respect to every processor on a multiprocessor.

Implementing uniprocessor locks by disabling interrupts

On a uniprocessor machine, any sequence of instructions by one thread will appear atomic to other threads if there is no context switch in the middle of the sequence. So, *on a uniprocessor machine* a thread can make a sequence of actions atomic by disabling interrupts (and refraining from calling thread library functions that can trigger a context switch) during the sequence.

This observation suggests a trivial—but seriously limited—approach to implementing locks on a uniprocessor machine:

```
Lock::acquire(){ disableInterrupts(); }
Lock::release(){ enableInterrupts(); }
```

This implementation does provide the mutual exclusion property we need from locks, and some uniprocessor kernels use this simple approach. However, it does not suffice as a general implementation for locks. If the code sequence the lock protects runs for a long time, interrupts will be disabled for a long time; this will prevent other threads from running for a long time, and it will make the system unresponsive to handling user inputs or other real-time tasks. Furthermore, although this approach can work within the kernel where all code is (presumably) carefully crafted and trusted to release the lock quickly, we cannot let untrusted user-level code run with interrupts turned off, since a malicious or buggy program could then monopolize a processor.

Implementing uniprocessor queuing locks

A more general solution is to briefly disable interrupts to protect the lock's data structures, but to reenable them once a thread has acquired the lock or determined that the lock is busy. In the latter case, there is no point in running the thread until the lock is free, so we suspend the thread by moving its thread control block (TCB) from the ready list to a queue of threads waiting for the lock.

```

class Lock{
    private:
        int value = FREE;
        Queue waiting;

    public:
        void Lock::Acquire(){
            disableInterrupts();
            if(value == BUSY){
                waiting.add(current thread's TCB); // This thread no longer on ready list
                suspend(); // Like yield(), but current thread's TCB
                           // is on waiting list rather than ready list
            }
            else {
                value = BUSY;
            }
            enableInterrupts();
        }

        void Lock::Release(){
            disableInterrupts();
            if(waiting.notEmpty()){
                move one TCB from waiting to ready;
            }
            else{
                value = FREE;
            }
            enableInterrupts();
        }
}

```

Figure 5.11: Pseudocode for uniprocessor lock implementation via disabling interrupts.

This approach is illustrated in the Lock implementation shown in Figure 5.11. In this pseudo-code, if a lock is BUSY when a thread tries to acquire it, the thread moves its thread control block (TCB) off of the ready list and onto the lock's waiting list; the thread then suspends itself using a call similar to `yield()`; since the thread's TCB is off the ready list, this `suspend()` call will not return until some calls `Lock::Release()` and moves the suspended thread's TCB to the ready list.

An optimization. Notice that if a thread is waiting for the lock, a call to `Lock::Release()` does not set `value` to FREE. Instead, it leaves `value` as BUSY. Then, the thread whose TCB is moved to the ready list is guaranteed to be the one that acquires the lock and returns next. This arrangement allows an implementation to ensure freedom from starvation: assuming that TCBs are removed from the `waiting` list in FIFO order, then a thread waiting for a lock is guaranteed to succeed in acquiring the lock within a bounded number of `Release()` calls on that lock.

This feature is an optimization of this specific implementation of the lock primitive. Users of locks should not assume anything about the order that waiting threads will acquire a lock.

Interrupt discipline and invariants. You may notice that a thread calls `suspend()` with interrupts turned off. Who turns them back on?

The next thread to run will turn interrupts back on. In particular, when we implement a thread system, we typically enforce the invariant that a thread always disables interrupts before performing a context switch and always assumes that interrupts are disabled when they run again after a context switch. So, whenever a thread returns from a context switch, it must reenable interrupts. For example, just as the `Lock::Acquire()` code in Figure 5.11 reenables interrupts before returning, the `yield()` code in Figure 4.12 on page 162 disables interrupts before the context switch and reenables interrupts before returning, and the interrupt handling approach described for context switches (Section 4.4.3) disables interrupts in the handler but restores interrupts when the interrupted thread is resumed.

Multiprocessor spinlocks

On a multiprocessor machine, however, disabling interrupts is insufficient. Even if interrupts are turned off, a sequence of operations by a thread on one processor can be interleaved with operations by another thread on another processor.

Atomic read-modify-write instructions. Since turning off interrupts is insufficient, most processor architectures provide *atomic read-modify-write* instructions to support synchronization. These instructions can read a value from a memory location to a register, modify the value, and write the modified value to memory atomically with respect to all instructions on other processors.

As an example, some architectures provide a `test_and_set` instruction, which atomically reads a value from memory to a register and writes a “1” to that memory location.

We can implement a spinlock that works on a multiprocessor (or a uniprocessor) using `test_and_set` as follows:

```
class SpinLock{
    private:
        int value = 0; // 0 = FREE; 1 = BUSY

    public:
        void SpinLock::Acquire(){
            while(test_and_set(&value)) // While busy
                ; // spin
        }

        void SpinLock::Release(){
            value = 0;
        }
}
```

Such a lock is called a `definitionspinlock` because a thread waiting for a BUSY lock “spins” in a tight

```

class Lock{
private:
    SpinLock spinlock;
    int value = FREE;
    Queue waiting;

public:
    void Lock::Acquire(){
        spinlock.acquire();
        if(value != FREE){
            disableInterrupts(); // Must finish what I start
            readyList->removeSelf(myTCB);
            waiting.add(myTCB);
            spinlock.release();
            suspend(); // Like yield(), but current thread's TCB
                        // is on waiting list or ready list
            enableInterrupts();
        }
        else{
            value = BUSY;
            spinlock.release();
        }
    }

    void Lock::Release(){
        spinlock.Acquire()
        if(waiting.notEmpty()){
            otherTCB = waiting.removeOne();
            readyList->add(otherTCB); // Ready list protected by
                                // its own spinlock
        }
        else{
            value = FREE;
        }
        spinlock.Release();
    }
}

```

Figure 5.12: Pseudo-code for a queuing lock that suspends threads that try to acquire the lock when it is BUSY.

loop until some other lock makes it FREE. This approach will be inefficient if locks are held during long operations on shared data, but if locks are known to only be held for short periods (i.e., less time than a context switch would take), spinlocks make sense. So, spinlocks are frequently used in multiprocessor kernels for shared objects whose methods all run fast.

Multiprocessor queuing locks

Often we need to support critical sections of varying length. For example, we may want a general solution to locks that does not make assumptions about the running time of methods that hold locks.

We cannot completely eliminate busy-waiting, but we can minimize it using the same approach as we did for uniprocessor locks based on disabling interrupts. Figure 5.12 shows an implementation of locks that uses a spinlock to guard

the (short) sequences of instructions that manipulate the lock's state but that suspends threads that try to acquire the lock when it is busy.

The basic idea is simple. A SpinLock called `spinlock` protects the lock's data structures. Once the `spinlock` is acquired a thread can inspect and update the lock's state. If a thread finds that the lock is `BUSY`, it removes itself from the ready list, adds itself to the lock's waiting list, and then suspends its execution to allow another thread to run. Later, when the lock is released, if any threads are waiting, then one of them is moved off the lock's waiting list to the scheduler's ready list.

Belt and suspenders. Why do we both acquire a spinlock and disable interrupts in the case where we suspend or resume a thread?

The basic issue is that we want to prevent a context switch during a context switch. Note that we must access two shared data structures—the lock and the scheduler's ready list. If the current thread is suspended after removing itself from the ready list but before adding itself to the lock's waiting list, then it will never be resumed. Turning off interrupts before beginning this sequence ensures that both operations occur before the thread is suspended.

Notice that turning off interrupts in this case does *not* ensure mutual exclusion. Instead, the lock and the ready list each have their own (spin-) locks to protect their own data structures.

Case study: Linux 2.6 kernel mutex lock

The pseudo-code above may seem a bit abstract, but real implementations closely follow this approach. For example, in Linux 2.6 the file `include/linux/mutex.h` defines a mutex's data structure as follows⁵:

```
struct mutex {
    /* 1: unlocked, 0: locked, negative: locked, possible waiters */
    atomic_t count;
    spinlock_t wait_lock;
    struct list_head wait_list;
};
```

This structure is similar to what we sketched in Figure 5.12. `wait_lock` and `wait_list` correspond to the `waiting` queue and spinlock's state; the `count` field is similar to the `value` field.

However, Linux provides an optimized path to acquire a free lock or release a lock with no waiting threads.

In particular, a lock can be in one of three main states as defined by `count`. If `count` is 1 the lock is unlocked. If `count` is 0, the lock is locked, but the

⁵For simplicity, throughout this example we include excerpts of the code and omit code for debugging code and for coordinating with signal handing, and we omit compiler/linker directives.

`wait_list` is empty. Finally, if `count` is negative, the lock is locked and there may be threads on the `wait_list`.

Note that to coordinate the fast and slow paths, the implementation uses atomic operations whenever it needs to manipulate `count`. Then, as long as a lock stays in the first two states, `lock()` and `unlock()` stay on their fast paths.

Acquiring the lock. To acquire a lock, a thread calls `mutex_lock()`, which is defined in `kernel/mutex.c`:

```
void mutex_lock(struct mutex *lock)
{
    __mutex_fastpath_lock(&lock->count, __mutex_lock_slowpath);
}
```

On a 32-bit x86 machine, `__mutex_fastpath_lock()` is defined in `arch/x_86/include/asm/mutex_32.h`:

```
/*
 * Change the count from 1 to a value lower than 1, and call <fn> if it
 * wasn't 1 originally. This function MUST leave the value lower than 1
 * even when the "1" assertion wasn't true.
 */
#define __mutex_fastpath_lock(count, fail_fn) \
do { \
    unsigned int dummy; \
    asm volatile(LOCK_PREFIX " decl (%eax)\n" \
                " jns 1f \n" \
                " call #fail_fn \n" \
                "1:\n" \
                : "=a" (dummy) \
                : "a" (count) \
                : "memory", "ecx", "edx"); \
} while (0)
```

The syntax of this inline assembly is a bit baroque, but the function itself is simple. Near the end, the notation `:="a" (count)` says that before running this assembly code, the x86 `eax` register should be initialized to hold the parameter `count`; notice from above that `count` holds the address of the mutex's `atomic_t count` field. Thus, the first x86 assembly instruction `LOCK_PREFIX decl(%eax)` is an atomic read-modify-write instruction that reads the old value of `count` from memory, decrements it, and stores the new value back to memory; the `LOCK_PREFIX` directive tells the assembler to use a version of the decrement instruction that executes atomically.

The second instruction `jns 1f` (“jump if not signed”) implements the fast path. If the new value of `count` is zero, then this conditional jump instruction jumps to the end of the assembly code snippet (to the `1:` label), and `__mutex_fastpath_lock()` returns. In this case, the lock is acquired in just two instructions!

On the other hand, if the new value of `count` is negative, the `jns` instruction falls through, and the `call #fail_fn` instruction calls the `__mutex_lock_slowpath()` function.

The slowpath function is implemented by `__mutex_lock_common()`, as described in the following excerpts from `kernel/mutex.c`.

```

/*
 * Lock a mutex (possibly interruptible), slowpath:
 */
static inline int __sched
__mutex_lock_common(struct mutex *lock)
{
    struct task_struct *task = current;
    struct mutex_waiter waiter;
    unsigned long flags;

    preempt_disable();
    spin_lock_mutex(&lock->wait_lock, flags);

    /* add waiting tasks to the end of the waitqueue (FIFO): */
    list_add_tail(&waiter.list, &lock->wait_list);
    waiter.task = task;
}

```

As shown in the excerpt above, the thread first disables interrupts, grabs the lock's `wait_lock` guard, and adds itself to the lock's `wait_list`.

Next in `__mutex_lock_common()` is the main loop:

```

for (;;) {
    /*
     * Lets try to take the lock again - this is needed even if
     * we get here for the first time (shortly after failing to
     * acquire the lock), to make sure that we get a wakeup once
     * it's unlocked. Later on, if we sleep, this is the
     * operation that gives us the lock. We xchg it to -1, so
     * that when we release the lock, we properly wake up the
     * other waiters:
     */
    if (atomic_xchg(&lock->count, -1) == 1)
        break;

    /* didnt get the lock, go to sleep: */
    spin_unlock_mutex(&lock->wait_lock, flags);
    preempt_enable_no_resched();
    schedule();
    preempt_disable();
    spin_lock_mutex(&lock->wait_lock, flags);
}

```

In this loop, the thread atomically swaps the value of the lock with -1 using an atomic read-modify-write `atomic_xchg` instruction. If the previous state was 1 (free), the lock is now owned by the calling thread, and it breaks out of the loop. Otherwise, the thread clears the `wait_lock` spinlock guard, moves itself off the ready queue, reenables interrupts (`preempt_enable_no_resched()`), and suspends its own execution to switch the processor to another thread (`schedule()`).

Later, when the thread runs again, it returns from `schedule()`, disables interrupts, and reacquires the lock's guard.

Eventually, the thread breaks out of the loop, which means that it found a moment when the lock was in the free state (the lock's `count` was 1), and at that moment it set the lock to the "busy, possible waiters" state (by setting `count = -1`.) The thread now has the lock, and it cleans up before exiting the acquire slow path as follows:

```
/* got the lock - rejoice! */
```

```

    mutex_remove_waiter(lock, &waiter, current_thread_info());

    /* set it to 0 if there are no waiters left: */
    if (likely(list_empty(&lock->wait_list)))
        atomic_set(&lock->count, 0);

    spin_unlock_mutex(&lock->wait_lock, flags);
    preempt_enable();

    return 0;
}

```

Here, the thread removes itself from the list of waiting threads. If there are no other waiting threads, it resets lock's state to "locked" (`count = 0`) so that when it releases the lock it can use the fast path if nothing has changed. Finally, the thread releases the lock's guard and reenables interrupts.

Releasing the lock. Lock release is similar. The function `mutex_unlock()` in `kernel/mutex.c` first tries a fast path and falls back on a slow path if needed:

```

void __sched mutex_unlock(struct mutex *lock)
{
    __mutex_fastpath_unlock(&lock->count, __mutex_unlock_slowpath);
}

```

And the fath path for unlock follows a similar path to that of lock. Here are the relevant excerpts from `arch/x86/include/asm/mutex_32.h`, for example:

```

#define __mutex_fastpath_unlock(count, fail_fn) \
do { \
    unsigned int dummy; \
    asm volatile(LOCK_PREFIX " incl (%eax)\n" \
                " jg 1f\n" \
                " call #fail_fn \n" \
                "1:\n" \
                : "=a" (dummy) \
                : "a" (count) \
                : "memory", "ecx", "edx"); \
} while (0)

```

Here, when we atomically increment `count`, if we find that the new value is 1, then the previous value must have been 0, so there can be no waiting threads and we're done. Otherwise, fall back to the slowpath by calling `fail_fn`, which corresponds to the following fuction in `kernel/mutex.c`.

```

static inline void
__mutex_unlock_common_slowpath(atomic_t *lock_count, int nested)
{
    struct mutex *lock = container_of(lock_count, struct mutex, count);
    unsigned long flags;

    spin_lock_mutex(&lock->wait_lock, flags);
    /*
     * Unlock lock here
     */
    atomic_set(&lock->count, 1);
    if (!list_empty(&lock->wait_list)) {
        struct mutex_waiter *waiter =
            list_entry(lock->wait_list.next,
                       struct mutex_waiter, list);

```

```

        wake_up_process(waiter->task);
    }
    spin_unlock_mutex(&lock->wait_lock, flags);
}

```

Notice that this function always sets `count` to 1, even if there are waiting threads. As a result, a new thread this is not waiting may swoop in and acquire the lock on its fast path, setting `count = 0`. However, in this case, the waiting thread will still eventually run, and when it does, the main loop above will set `count = -1`.

Discussion: Mutex fast path. An important thing to remember from this example is that many implementations of locks include a fast path so that acquiring and releasing an uncontended lock is cheap. Programmers will sometimes go to great lengths to convince themselves that they can avoid acquiring a lock in a particular situation. However, the reasoning in such cases can be subtle, and omitting needed locks is dangerous. In cases where there is little contention, avoiding locks is unlikely to significantly improve performance, so it is usually better just to keep things simple and rely on locks to ensure mutual exclusion when accessing shared state.

Locks for user-level and kernel-supported threads

The discussion above focused on implementing locks for in-kernel threads. In that case everything—code, shared state, lock data structures, thread control blocks, and the ready list—is in kernel memory, and all threads run in kernel mode. Fortunately, although some details change, the basic approaches work when we implement locks for threads that run within user-level processes.

Kernel-supported threads. In a kernel-supported threads library, the kernel provides multiple threads to a process. In this situation, the kernel scheduler needs to know when a thread is waiting for the lock so that it can suspend the waiting thread and run a different one.

In the simplest case, we can place the lock data structure, including all of a lock’s state (e.g., `value`, `waiting`, and the `spinlock` in the kernel’s address space and replace each method call on the lock object with a system call. Then, the implementations described above for kernel-level locks can be used without significant changes.

A more sophisticated approach splits the lock’s state and implementation into a fast path and slow path similar to the Linux lock described above. For example, each lock has two data structures: one in the process’s address space that holds something similar to the `count` field and one in the kernel with the `spinlock` and `wait_list` queue.

Then, acquiring a free lock or releasing a lock that has no waiting threads can be done with a few instructions executed by the user-level thread and without a system call. A system call is still needed when the fast path fails (e.g., when the thread needs to remove itself from the ready list, add itself to the waiting list, and suspend execution.) We leave the details of such an implementation as an exercise for the reader.

User-level threads. In a threads library that operates completely at user level, the kernel provides just one thread to a process, and the process multiplexes user-level threads over that one virtual processor. The situation is similar to the kernel-level threads case, except everything operates in a process's address space rather than in the kernel's address space. In particular, all of the code, shared state, lock data structures, thread control blocks, and the ready list are in the process's address space.

Lock implementations based on disabling interrupts change only slightly. A user-level threads package cannot disable interrupts; the kernel can not allow an untrusted process to disable interrupts and potentially run forever. Instead, it can disable signals from the operating system, which effectively accomplishes the same thing: just as a thread in a kernel-level threads package running on a uniprocessor machine can make a set of operations atomic by disabling interrupts before the operations and enabling them at the end, a thread in a user-level threads package that multiplexes user-level threads over one kernel thread can make a set of operations atomic by disabling signals before the operations and disabling them at the end.

Implementations based on atomic read-modify-write instructions need no changes at all.

5.5.2 Implementing condition variables

We can implement condition variables using the same techniques we use to implement locks.

To illustrate the similarity, Figure ?? shows a implementation of condition variables for a kernel-level threads based on atomic read-modify-write instructions. There are few changes from the lock implementation for that environment shown in Figure 5.12 on page 216. For example, as in the lock implementation, we still disable interrupts to ensure that once a thread removes itself from the ready list it puts itself onto the waiting list.

This implementation provides Hansen semantics—when we signal a waiting thread, that thread becomes READY, but it may not run immediately, and it still must reacquire the lock. It is possible for another thread to acquire the lock first and to change the state guarded by the lock before the waiting thread returns from `Cond::Wait()`.

```

class Cond{
    private:
        SpinLock spinlock;
        Queue waiting;

    public:
        void Cond::Wait(Lock *lock){
            spinlock.Acquire();
            disableInterrupts();           // Must finish what I start
            readyList->removeSelf(myTCB);
            waiting.add(myTCB);
            lock->Release();
            spinlock.Release();
            suspend(); // Like yield(), but current thread's TCB
                        // may be on the waiting list or the ready list
            enableInterrupts();
            lock->Acquire();
        }

        void Cond::Signal(){
            disableInterrupts();
            if(waiting.notEmpty()){
                move one TCB from waiting to ready;
            }
            enableInterrupts();
        }

        void Cond::Broadcast(){
            disableInterrupts();
            while(waiting.notEmpty()){
                move all TCBs from waiting to ready;
            }
            enableInterrupts();
        }
}

```

Figure 5.13: Pseudocode for condition variable implementation via atomic read-modify-write instructions.

Exercises

7. Wikipedia provides an implementation of Peterson's algorithm to provide mutual exclusion using loads and stores at [http://en.wikipedia.org/wiki/Peterson's_algorithm](http://en.wikipedia.org/wiki/Peterson%27s_algorithm). Unfortunately, this code is not guaranteed to work with modern compilers or hardware. Update the code to include memory barriers where necessary. (Of course you could add a memory barrier before and after each instruction; your solution should instead add memory barriers only where necessary for correctness.)
8. Linux provides a `sys_futex()` system call to assist in implementing hybrid user-level/kernel-level locks and condition variables.

A call to `long sys_futex(void *addr1, FUTEX_WAIT, int val1, NULL, NULL, 0` checks to see if the memory at address `addr1` has the same value as `val1`. If so, the calling thread is suspended. If not, the calling thread returns immediately with the error return value `EWOULDBLOCK`. In addition,

the system call will return with the value EINTR if the thread receives a signal.

A call to `long sys_futex(void *addr1, FUTEX_WAKE, 1, NULL, NULL, 0)` causes one thread waiting on `addr1` to return.

Consider the following (too) simple implementation of a hybrid user-level/kernel-level lock.

```
class TooSimpleFutexLock{
    private:
        int val;

    public:

        TooSimpleMutex() : val(0) {} // Constructor

        void Acquire() {
            int c;
            while ((c = atomic_inc(&val)) != 0){ // atomic_inc returns *old* value
                futex_wait(&val, c + 1);
            }
        }

        void Release() {
            val = 0;
            futex_wake(&val, 1);
        }
};
```

There are three problems with this code.

- (a.) **Performance.** The goal of this code is to avoid making (expensive) system calls in the uncontested case when an `Acquire()` tries to acquire a free lock or a `Release()` call releases a lock with no other waiting threads. This code fails to meet this goal. Why?
- (b.) **Performance.** There is a subtle corner case when multiple threads try to acquire the lock at the same time that can show up as occasional slowdowns and bursts of CPU usage. What is the problem?
- (c.) **Correctness.** There is a corner case that can cause the mutual exclusion correctness condition to be violated, allowing two threads to both believe they hold the lock. What is the problem?

5.6 Designing and implementing shared objects

Although multi-threaded programming has a reputation for being difficult, shared objects provide a basis for writing simple, safe code for multi-threaded programs. In this section, we first define a high-level approach to designing shared objects. Then, we define some specific rules that you should follow when you implement them. Our experience is that following this approach and these rules makes it

much more likely that you will write code that is not only correct but also easy for others to read, understand, and maintain.

Of course, writing individual shared objects is not enough. Most programs have multiple shared objects, and new issues arise when combining them. But, before trying to compose multiple shared objects, we must make sure that each individual object works. Chapter ?? discusses the issues that arise when programs use multiple shared objects.

5.6.1 High level design

In the discussion above, observe that a shared object has public methods, private methods, state variables, and synchronization variables, where a shared object's synchronization variables are a lock and one or more condition variables. At this level, shared object programming is basically like standard object oriented programming, except we've added synchronization variables to each shared object. This similarity is deliberate: synchronization variables are carefully defined so that we can continue to apply familiar techniques for programming and reasoning about objects.

So, most of high level design challenges for a shared object's class are the same as for class design in single-threaded programming:

- Decompose the problem into objects
- For each object
 - Define a clean interface
 - Identify the right internal state and invariants to support that interface
 - Implement methods with appropriate algorithms to manipulate that state.

All of these steps require creativity and good engineering judgement and intuition. Going from single-threaded to multi-threaded programming does not make these steps particularly more difficult.

Compared to what you do to implement a class in a single-threaded program, the new steps needed for the multi-threaded case for shared objects are straightforward.

- Add a lock
- Add code to acquire and release the lock
- Identify and add condition variables

- Add loops to wait using the condition variable(s)
- Add `signal()` and `broadcast()` calls

We will talk about each of these issues in turn.

Other than these fairly-mechanical changes, writing the rest of the code for your proceeds as it does in the single-threaded case.

5.6.2 Add a lock

Add a lock as a member variable for each object in the class to enforce mutual exclusion on access to each object's shared state.

Note that in this chapter, we focus on the simple case where each shared object includes exactly one lock. Later, we will talk about more advanced variations such as an ownership design pattern where higher-level program structure enforces mutual exclusion by ensuring that at most one thread at a time owns and can access an object and fine grained locking where a single object may subdivide its state into multiple parts, each protected by its own lock.

5.6.3 Acquiring and releasing the lock

All code that accesses the object's state that is shared across more than one thread must hold the object's lock. Typically, all of an object's member variables are shared state.

The simplest and most common thing to do is to acquire the lock at the start of each public method and release it at the end of each public method. If you do this, it is easy to inspect your code to verify that a lock is always held when needed. Also, if you do this, then the lock is already held when each private method is called, and you don't need to reacquire it.

Warning. You may be tempted to try to avoid acquiring the lock in some methods or some parts of some methods. Do not be tempted by this “optimization” until you are a very experienced programmer and have done sufficient profiling of the code to verify that the optimization you are considering will significantly speed up your program.

Remember that acquiring an uncontended lock is a relatively inexpensive operation. Also, remember from the Too Much Milk problem that reasoning about memory interleavings can be quite difficult—and the instruction reordering done by modern compilers and processors makes it even harder. The sidebar discusses one commonly used (and abused) “optimization.”

5.6.4 Identifying condition variables

How should a programmer decide what condition variables a shared object needs?

A systematic way to approach this problem is to consider each method and ask, “*When can this method wait?*” Then, you can map each situation in which a method can wait to a condition variable.

A programmer has considerable freedom in deciding how many condition variables a class should have and what each should represent. One option is to add a condition variable corresponding to each situation in which a call to the method must wait—perhaps creating a distinct condition variable for each method that can block.

Example: Blocking Bounded Queue with two condition variables. In our blocking bounded queue example, if the queue is full `insert()` must wait until someone removes an item, so we create a condition variable `itemRemoved`. Similarly, if the queue is empty `remove()` must wait until someone inserts an item, so we create a condition variable `itemAdded`. It is fairly natural in this case to create two condition variables, `itemAdded` to wait until the queue has items and `itemRemoved` to wait until the queue has space.

Alternatively, a single condition variable can often suffice. In fact, early versions of Java defined a single condition variable per object and did not allow programmers to allocate additional ones, so this approach was effectively mandated by the language. Under this approach, any thread that waits for any reason uses that condition variable; if the condition variable is used by different threads waiting for different reasons, then any thread that makes a state change that could allow one or more other threads to proceed broadcasts on that variable.

Example: Blocking Bounded Queue with one condition variable. It is also possible to implement the blocking bounded queue with a single condition variable `somethingChanged` on which both threads in `insert()` or threads in `remove()` can wait. If we choose this approach, `insert()` and `remove()` will need to `Broadcast()` rather than `Signal()` to ensure that the right threads get a chance to run.

More complex programs make these trade-offs more interesting.

Example: ResourceManager. For example, imagine a `ResourceManager` class that allows a calling thread to request exclusive access of any subset of n distinct resources. One could imagine creating 2^n condition variables so that a requesting thread could wait on a condition variable representing exactly its desired combination and the signaling on all affected condition variables when a resource becomes free. However, it may be simpler in this case to have a single condition variable on which requesting threads wait and to broadcast on that condition whenever a resource is freed.

The bottom line here is that there is no hard and fast rule for how many condition variables to use in a shared object or selecting the conditions that map to each condition variable. Selecting condition variables requires some thought, and different designers may group blocking situations differently and end up with different condition variables for a class. Like many other design decisions, these decisions are a matter of programmer taste, judgement, and experience. The point is that asking “When can this method wait” may help you identify what is for you a natural way of thinking about the condition variables for a shared object.

5.6.5 Waiting using condition variables

Add a `while(...){cv.Wait()}` loop into each method that you just identified as potentially needing to wait before returning.

Remember that every call to `Condition::Wait()` must be enclosed in a while loop that tests an appropriate predicate—modern implementations almost invariably enforce Hansen semantics and often allow for spurious wake-ups (a thread can return from `Wait()` even if no thread called `Signal()` or `Broadcast()`). Therefore, a thread must always check the condition before proceeding—even if the condition was true when the `Signal()` or `Broadcast()` call occurred, it may no longer be true when the waiting thread resumes execution.

Modularity benefits. Notice that if you always wait in a while loop, your code becomes highly modular. You can look at the code that waits and know what is true when it proceeds *without* examining any other code or understanding when calls to `Signal()` or `Broadcast()` are made. Even erroneous calls to `Signal()` or `Broadcast()` will not change how the waiting code behaves.

For example, consider the assertion in the following code:

```
...
while(!workAvailable()){
    cond.wait(&lock);
}
assert(workAvailable());
...
```

We know that the assertion holds by local inspection *without knowing anything about the more distant code that calls Signal() or Broadcast()*.

Waiting in a while loop also makes the signal and broadcast code more robust. Adding an extra `Signal()` or changing a `Signal()` to a `Broadcast()` will not introduce bugs.

Hint: Top-down design. As you start writing your code, you may know that a method needs to include a wait loop, but you may not know exactly

what the predicate should be. In this situation, we often find it useful to name a private method function that will perform the test (e.g., `workAvailable()` in the above example) and write the code that defines the details for that test later.

5.6.6 `Signal()` and `Broadcast()` calls

Just as you must decide when methods can wait, you must decide when methods can let other waiting threads proceed. It is usually easy to ask “Can a call to this method allow another thread to proceed?” and then add a `Signal()` or `Broadcast()` call if the answer is yes. But which call should you use?

`Signal()` is appropriate when (1) at most one waiting thread can make progress and (2) any thread waiting on the condition variable can make progress. In contrast, `Broadcast()` is needed when (1) multiple waiting threads may all be able to make progress or (2) different threads are using the same condition variable to wait for different predicates, so some of the waiting threads can make progress but others can’t.

Example: Resource Manager. As an example of the latter case, suppose that we use a single condition variable the *n*-resource `ResourceManager` sketched above. Whenever a resource is freed, we must `broadcast()` on the condition variable: we don’t know which thread(s) can make progress, so we tell them all to check. If, instead, we just signalled, then the “wrong” thread might receive the signal, and a thread that could make progress might remain blocked.

It is always safe to use `Broadcast()`. Even in cases where signal would suffice, at worst all of the waiting threads would run and check the condition in the `while` loop, but only one would continue out of the loop. Compared to `signal()`, such a case consumes some additional resources, but would not introduce any bugs.

Double Checked Locking

The body of the text advises holding a shared object’s lock across any method that accesses the object’s member variables. Programmers are often tempted to avoid some of these lock acquire and release operations. Unfortunately, such efforts often result in code that is complex, wrong, or both.

To illustrate the challenges, consider the *double checked locking* design pattern. The canonical example⁶ is for an object `Singleton` that provides access to an object that is allocated lazily the first time it is needed by any thread.

⁶The example and analysis is taken from Meyers and Alexandrescu’s “C++ and the Perils of Double-Checked Locking”

The “optimization” is to avoid acquiring the lock if the object has not already been allocated, but to avoid allocating the object multiple times by acquiring the lock and rechecking the status before allocating the object:

```
Singleton* Singleton::pInstance = NULL;
// BUG! Don't do this!
Singleton* Singleton::instance() {
    if (pInstance == NULL) {
        lock->Acquire();
        if (pInstance == NULL){
            pInstance = new Instance();
        }
        lock->Release();
    }
    return pInstance;
}
```

Singleton.h header file

Singleton.cc implementation file

Although the intuition is appealing, this code does not work. The problem is that the statement `pInstance = new Instance()` is not an atomic operation; in fact, it comprises at least three steps:

1. Allocate memory for a `Singleton` object
2. Initialize the `Singleton` object’s memory by running the constructor
3. Make `pInstance` point to this newly constructed object

The problem is that, modern compilers and hardware architectures can reorder these events. Thus, it is possible for thread 1 to execute the first step and then the third step; then thread 2 can call `instance()`, see that `pInstance` is non-null, return it, and begin using this object before thread 1 initializes it.

Discussion. This is just an example of dangers that lurk when you try to elide locks; the lesson applies more broadly. This example is extremely simple—fewer than 10 lines of code with very simple logic—yet a number of published solutions have been wrong. For example, Meyers and Alexandrescu’s “C++ and the Perils of Double-Checked Locking” notes, some tempting solutions using temporary variables and the `volatile` keyword don’t work.⁷ Bacon et al.’s “The ‘Double-Checked Locking is Broken’ Declaration” discusses a wide range of non-working solutions in Java.

This type of optimization is risky and often does not end up providing significant performance gains in programs. Most programmers should not consider them. Even expert programmers should habitually stick to simpler programming patterns like the ones we discuss in the body of the text and should consider optimizations like double-checked locking only rarely and only when performance measurements and profiling indicate that the optimizations will matter for overall performance.

⁷There are (non-portable) solutions in C/C++, but we won’t give them here. If you want to try these advanced techniques, then you should read the literature for more in-depth discussions so that you deeply understand the issues and why various appealing “solutions” fail.

5.6.7 Implementation best practices: Writing simple, safe code with shared objects

Above we described the basic thought process you will follow when including and using locks and condition variables in a shared object. To make things more concrete, this section describes five simple rules that we strongly advocate as a set of best practices for writing code for shared objects.

Consistent structure. At the core of many of these rules is a simple principle: *follow a consistent structure*. This is a meta-rule that underlies the other rules. Although programming with a clean, consistent structure is always useful, it is particularly important to strictly follow tried and true design patterns for shared objects.

At a minimum, even if one way is not better than another, following the same strategy every time (a) frees you to focus on the core problem because the details of the standard approach become a habit and (b) makes it easier for the people who follow you and have to review, maintain, or debug your code understand your code. (Of course, often the person that has to debug the code is you!)

As an analogy, electricians follow standards for the colors of wire they use for different tasks. White is neutral. Black or red is hot. Copper is ground. An electrician doesn't have to decide "Hm. I have a bit more white on my belt today than black, should I use white or black for my grounds?" When an electrician walks into a room she wired last month, she doesn't have to spend 2 minutes trying to remember which color is which. If an electrician walks into a room she has never seen before, she can immediately figure out what the wiring is doing (without having to trace it back into the switchboard.) Similar advantages apply to following coding standards.

However, for concurrency programs, the evidence is that in fact the abstractions we describe *are* better than almost all others. Until you become a *very* experienced concurrent programmer, you should take advantage of the hard-won experience of those that have come before you. Once you are a concurrency guru, you are welcome to invent a better mousetrap.

Sure, you can cut corners and occasionally save a line or two of typing by departing from the standards, but you'll have to spend a few minutes thinking to convince yourself that you are right on a case-by-case basis (and another few minutes typing comments to convince the next person to look at the code that you're right), and a few hours or weeks tracking down bugs when you're wrong. It's just not worth it.

Five rules. These five rules are designed to help you avoid common pitfalls we see in multi-threaded code from students and experienced programmers.

Coding standards, soap boxes, and preaching

Some people rebel against coding standards. We do not understand the logic. For concurrent programming in particular, there are a few good solutions that have stood the test of time (and many unhappy people who have departed from these solutions.) For concurrent programming, *debugging will not work*. You must rely on (a) writing correct code and (b) writing code that you and others can read and understand—not just for now, but also over time as the code changes. Following the rules below will help you write correct, readable code.

When we teach multithreaded programming, we treat the six rules described in this section as *required coding standards* for all multi-threaded code that students write in the course. We say, “We can’t control what you do when you leave this class, but while you are in this class, any solution that violates these standards is, by definition, *wrong*.”

In fact, we feel so strongly about these rules that one of us actually presents them in class by standing on a table and pronouncing these as the Six Commandments of multi-threaded programming:

1. Thou shalt always do things the same way

etc.

The particular formulation (and presentation) of these rules evolved from our experience teaching multi-threaded programming dozens of times to hundreds of students and identifying common mistakes. We have found that when we insist that students follow these rules (adding a bit of drama to drive home the point), the vast majority find it easy to write clear and correct code for shared objects. Conversely, in earlier versions of the course when we were more subtle and phrased these items as “strong suggestions,” many students found themselves adrift, unable to write code for even the simplest shared objects.

Our advice to those learning multi-threaded programming is to treat these rules as a given and follow them strictly for a semester or so, until writing shared objects is easy. At that point, you understand things well enough to decide whether to continue to follow them.

We also believe that experienced programmers benefit from adhering closely to these rules. Since we began teaching these rules in their current form, we have also disciplined ourselves to follow them unless there is a very good reason not to. We have found exceptions few and far between. Conversely, the vast majority of the time when we catch ourselves being tempted to deviate from the rules and then force ourselves to rewrite the code to follow them, the code improves.

We should note that although these rules may come across as opinionated (and they are), they are far from novel. For example, over three decades ago, Lampson and Redell’s paper “Experience with Processes and Monitors in Mesa” (*Communications of the ACM* v. 23 n. 2 Feb 1980) provided similar advice (albeit in a much more measured tone.)

1. Always synchronize with locks and condition variables.

Either locks and condition variables or semaphores could be used to implement shared objects. We recommend that you be able to read and understand semaphores so you can understand legacy code, but that you only write new code using locks and condition variables.

99% of the time code with locks and condition variables is more clear than the equivalent semaphore code because it is more “self-documenting”. If the code is well-structured, it is usually clear what each synchronization action is doing. Admittedly, occasionally semaphores seem to fit what you are doing perfectly because you can map the object’s invariants onto the internal state of the semaphore exactly—for example you can write an extremely concise version of our blocking bounded queue using semaphores—but what happens when the code changes a bit next month? Will the fit be as good? For consistency and simplicity, you should choose one of the two styles and stick with it, and in our opinion, the right one to pick is to use locks and condition variables.

2. Always acquire the lock at the beginning of a method and release it right before the return.

This is mainly an extension of the principle of consistent structure: pick one way of doing things and always follow it. The benefit here is that it is easy to read code and see where the lock is held and where it isn’t because synchronization is structured on a method-by-method basis. Conversely, if `Lock::Acquire()` and `Lock::Release()` calls are buried in the middle of a method, it is harder to quickly inspect and understand the code.

Taking a step back, if there is a logical chunk of code that you can identify as a set of actions that require a lock, then that section should probably be its own procedure—it is a set of logically related actions. If you find yourself wanting to grab a lock in the middle of a procedure, that is usually a red flag that you should break the piece you are considering into a separate procedure. We are all sometimes lazy about creating new procedures when we should. Take advantage of this signal, and you will write clearer code.

3. Always hold the lock when operating on a condition variable.

The reason you signal on a condition variable because you just got done manipulating shared state—some other thread is waiting in a loop for some test on shared state become to true. Condition variables are useless without shared state and shared state should only be accessed while holding a lock.

Many libraries enforce this rule—you cannot call any condition variable methods unless you hold the corresponding lock. But some run-time systems and libraries allow sloppiness, so take care.

4. Always wait in a `while()` loop

E.g., the pattern is always

```
while(predicateOnStateVariables(...)){
    condition->Wait(&lock);
}
```

never

```
if(predicateOnStateVariables(...)){
    condition->Wait(&lock);
}
```

Here, `predicateOnStateVariables(...)` is code that looks at the state variables of the current object to decide if it is OK to proceed.

One is sometimes tempted to guard a `wait()` call with an `if` conditional rather than a `while` loop when one knows exactly what threads are taking what actions and one can deduce from the global structure of the program that despite Hansen semantics, any time a thread returns from `wait()`, it can proceed. Avoid this temptation.

`While` works any time `if` does, and it works in situations when `if` doesn't. By the principle of consistent structure, you should do things the same way every time. But there are three additional issues.

First, `if` breaks modularity. In the example sketched above, one needs to consider the global structure of the program—what threads are there, where is `signal()` called, etc—to determine whether `if` will work. The problem is that a change in code in one method (say, adding a `signal()`) can then cause a bug in another method (where the `wait()` is). `While` code is also self-documenting—one can look at the `wait()` and see exactly when a thread may proceed.

Second, when you always use `while`, you are given incredible freedom about where you put the `signals()`. In fact, `signal()` becomes a hint—you can add more signals to a correct program in arbitrary places and it remains a correct program.

Third, `if` breaks portability. Some implementations of condition variables may allow *spurious wakeups*, where `wait()` returns even though no thread called `signal()` or `broadcast()`. For example, implementations of condition variables in both Java and the Posix pthreads library are allowed to have spurious wakeups.

5. (Almost) never `sleep()`.

Many threads libraries have a `sleep()` function that suspends execution of the calling thread for some period of wall clock time. Once that time passes, the thread is returned to the scheduler's ready queue and can run again.

Never use `sleep()` to have one thread wait for another thread to do something. The correct way to wait for a condition to become true is to `wait()` on a condition variable.

In general, `sleep()` is only appropriate when there is a particular real-time moment when you want to perform some action. If you catch yourself writing `while(testOnObjectState() {sleep();};`, treat this as a big red flag that you are probably making a mistake.

Similarly, if a thread needs to wait for an object's state to change, it should wait on a condition variable, not just call `yield()`. Instead, `yield()` is appropriate when a low priority thread that could make progress instead yields the processor to let higher-priority threads run.

Two pitfalls in Java

Java is a modern type-safe language that included support for threads from its inception. This built-in support makes multi-threaded programming in Java convenient. However, some aspects of the language are *too* flexible and can encourage bad practices. We highlight two pitfalls here.

1. Avoid defining a synchronized block in the middle of a method

Java provides built in language support for shared objects (“monitors.”) The base `Object` class, from which all classes inherit, includes a lock and a condition variable as members. Then, any method declaration can include the keyword `synchronized` to indicate that the object's lock is to be automatically acquired on entry to the method and automatically released on any return from the method. E.g.,

```
public synchronized foo(){  
    // Do something; lock is automatically acquired/released.  
}
```

This syntax is useful—it follows rule #2 above, and it frees the programmer from having to worry about details like making sure the lock is released before every possible return point including exceptions. The pitfall is that Java also allows a *synchronized block* in the middle of a method. E.g.,

```
public bar(){  
    // Do something without holding the lock  
    synchronized{  
        // Do something while holding the lock  
    }  
    // Do something without holding the lock  
}
```

This construct violates rule #2 from Section 5.6.7 and suffers from the disadvantages listed there. The solution is the same as discussed above: when you find yourself tempted to write a synchronized block in the middle of a Java method, treat that as a strong hint that you should define a separate method to more clearly encapsulate the logical chunk you have identified.

2. Keep *shared state classes* separate from *thread classes*

Java defines a class called `Thread` that implements an interface called `Runnable` that other classes can implement in order to be treated as threads by the runtime system. To write the code that represents a thread's “main loop”, you typically extend the `Thread` class or implement a class that implements `Runnable`.

The pitfall is that when extending the `Thread` class (or writing a new class that implements `Runnable`) programmers can be tempted to include not only the thread's main loop but also state that will be shared across multiple threads, blurring the lines between the threads and the shared objects. This is almost always confusing.

For example, for a blocking bounded queue, rather than defining two classes, `BBQ` for the shared queue and `WorkerThread` for the threads, programmers are sometimes tempted to combine the two into a single class—for example, a queue with an associated worker thread. If this sounds confusing, it is, but it is a pitfall we frequently see among students.

The solution is simple—always make sure threads and shared objects are defined in separate classes. State that can be accessed by multiple threads, locks, and condition variables should never appear in any Java class that extends `Thread` or implements `Runnable`.

5.6.8 Example: Readers/writers

Multithreaded programming is an important skill, and we anticipate that almost everyone who reads this book will be frequently called on to write multithreaded programs. This section and the next walk through two substantial examples that illustrate how to implement shared objects.

First, we will solve the classic *readers/writers* problem. In this problem, we have a database that can have records that can be read and written. To maximize performance, we will allow multiple threads to simultaneously read a record. However, for correctness, if any thread is writing a record, no other thread may simultaneously read or write that record.

Thus, we need to generalize our mutual exclusion Lock into readers-writers lock. We will do this by implementing a new kind of a new kind of shared object, `RWLock`, to guard access to each record and enforce these rules. The `RWLock` will be implemented using our standard synchronization building blocks: mutual exclusion locks and condition variables.

Then, a thread that wants to read a record will proceed as follows:

```
rwLock->startRead();
// Read database entry
rwLock->doneRead();
```

Similarly, a thread that wants to write a record will do the following:

```
rwLock->startWrite();
// Write database entry
rwLock->doneWrite();
```

```

class RWLock{
private:
    // Synchronization variables
    Lock lock;
    Cond readGo;
    Cond writeGo;

    // State variables
    int activeReaders;
    int activeWriters;
    int waitingReaders;
    int waitingWriters;

public:
    RWLock();
    ~RWLock() {};
    void startRead();
    void doneRead();
    void startWrite();
    void doneWrite();

private:
    bool readShouldWait();
    bool writeShouldWait();
};

```

Figure 5.14: `RWLock.h` defines the interface and member variables for our readers/writers solution.

To design a class, we begin by defining its interface (already done in this case) and the state needed for the interface. For the latter, it is important to keep enough state in the shared object to allow a precise characterization of the state—it is usually better to have too much state than to little. Here, the object’s behavior is fully characterized by the number of threads reading or writing and the number of threads waiting to read or write, so we have chosen to keep four integers to track these values. Figure 5.14 shows the members of and interface to the `RWLock` class.

Next, we add synchronization variables by asking “When can methods wait?” First, we add a mutual exclusion lock since a method must wait if another thread is accessing shared state. Next, we observe that `startRead()` or `startWrite()` may have to wait, so we add a condition variable for each case: `readGo` and `writeGo`.

`DoneRead()` and `doneWrite()` do not wait (other than to acquire the mutual exclusion lock), so these methods do not suggest the need for any additional condition variables.

We can now implement `RWLock`. Figure 5.15 shows the complete solution, which we develop in a few simple steps.

Much of what we need to do is almost automatic.

- Since we always acquire/release mutual exclusion locks at the beginning/end

of a method (never in the middle), we can write calls to acquire and release the mutual exclusion lock at the start and end of each public method before even thinking in detail about what these methods do.

At this point, `startRead()` and `doneRead()` look like this:

```
void RWLock::startRead()
{
    lock.Acquire();
    lock.Release()
}

void RWLock::doneRead()
{
    lock.Acquire();
    lock.Release()
}
```

`startWrite()` and `doneWrite()` are similar. So far, we can write all of these without much thought.

- Since we know `startRead()` and `startWrite()` may have to wait, we can write a `while(...){wait(...);}` loop in the middle of each. In fact, we can defer thinking too hard about the details by making the predicate for the while loop be a call to a private method that checks a condition to be defined later (e.g., `readShouldWait()` and `writeShouldWait()`).

At this point, `startRead()` looks like this:

```
void RWLock::startRead()
{
    lock.Acquire();
    while(readShouldWait()){
        readGo.Wait(&lock);
    }
    lock.Release();
}
```

`StartWrite()` looks similar.

Now we do need to think a bit. We can add code to maintain the invariants that `activeReaders`, `activeWriters`, `waitingReaders`, and `waitingWriters` track the state of the threads as expected from their names; since we hold mutual exclusion locks in all of the public methods, this is easy to do. For example, a call to `startRead()` initially increments the number of waiting readers; then when the thread gets past the `while` loop, the number of waiting readers is decremented but the number of active readers is incremented.

When reads or writes finish, it may become possible for waiting threads to proceed. We therefore need to add some `signal()` or `broadcast()` calls to `doneRead()` and `doneWrite()`. The easiest thing to do would be to broadcast on both `readGo` and `writeGo` in each method, but that would both be inefficient and (to our taste) be less clear about how the class actually works.

Instead, we observe that in `doneRead()` when a read completes, there are two interesting cases: (a) no writes are pending and nothing needs to be done

since this read cannot prevent other reads from proceeding or (b) a write is pending and this is the last active read, so one write can proceed. In case (b) we signal since at most one write can proceed and since any write waiting on the condition variable can proceed.

Our code for `startRead()` and `doneRead()` is now done:

```
void RWLock::startRead()
{
    lock.Acquire();
    waitingReaders++;
    while(readShouldWait()){
        readGo.Wait(&lock);
    }
    waitingReaders--;
    activeReaders++;
    lock.Release();
}

void RWLock::doneRead()
{
    lock.Acquire();
    activeReaders--;
    if(activeReaders == 0
       && waitingWriters > 0){
        writeGo.Signal(&lock);
    }
    lock.Release();
}
```

Code for `startWrite()` and `endWrite()` is similar. As Figure 5.15 indicates, for `doneWrite()`, if there are any pending writes, we signal on `writeGo`. Otherwise, we broadcast on `readGo`.

All that is left is to define the `readShouldWait()` and `writeShouldWait()` predicates. Here, we implement what is called a *writers preferred* solution: reads should wait if there are any active or pending writes, and writes should wait if there are any active reads or writes. Otherwise, a continuous stream of new reads could starve a write request and prevent it from ever being serviced.

```
bool
RWLock::readShouldWait()
{
    if(activeWriters > 0 || waitingWriters > 0){
        return true;
    }
    return false;
}
```

The code for `writeShouldWait()` is similar.

Notice that since `readShouldWait()` and `writeShouldWait()` are private methods that are always called from public methods that hold the mutual exclusion lock, they do not need to acquire the lock.

Our solution may not be to your taste. You may decide to use more or fewer condition variables, use different state variables to implement different invariants, or tweak when we `Signal()` or `Broadcast()`. The shared object approach allows designers freedom in these dimensions.

5.6.9 Example: The sleeping barber

Versions of the Sleeping Barber problem date back to at least Dijkstra in 1965 (“Cooperating sequential processes,” EWD 123, 1965, <http://userweb.cs.>

utexas.edu/users/EWD/transcriptions/EWD01xx/EWD123.html). The Sleeping Barber problem is illustrative of a system where several threads need to rendezvous with another thread that provides some service to them (e.g., several document editor threads printing documents via a printer driver thread.)

In our version, there is a barbershop with one barber chair and a waiting room with `NCHAIRS` chairs. The barber arrives every morning. When the shop is open, while there are customers in the shop, the barber cuts one customer's hair at a time. While the shop is empty of customers, the barber naps in the barber chair. At closing time, a clock rings an alarm, the barber hangs a "closed" sign, cuts the hair of anyone already in the shop, and then departs when the shop is empty.

When a customer arrives at the shop, if the shop is closed, the customer departs. Otherwise, the customer enters the shop. Then, if the barber is asleep, the customer wakes the barber, if there is a free waiting room chair, the customer sits down to wait, but if all waiting room chairs are full, the customer leaves. Waiting customers are served in FIFO order.

Figures 5.16 and 5.17 show our solution.

We represent the barber shop as a shared object `BarberShop` and the barber, quitting-time clock, and each customer as a thread. The barber will call `BarberShop::barberDay()` to open the shop. Each customer will call `BarberShop::getHairCut()`, which returns true if the customer succeeds in getting a haircut or false if the customer fails because the shop is closed or full. Finally, at some point the clock thread will call `BarberShop::clockRingsClosingTime()` to tell the barber to close shop.

As you can see in Figure 5.17, a barber's day is not an atomic action. In particular, customers can continue to arrive while the barber is cutting hair (we're using the Sleeping Barber problem to model a system where we have a worker thread doing work that arrives and lands in a queue.) So, rather than holding the lock across `barberDay()`, we break `barberDay()` into three pieces, `openStore()`, `waitForCustomer()`, and `doneCutting()`, each of which (as per rule #2) holds the lock from start to finish. The barber "cuts hair" by printing a message using `printf()`; this step represents IO or some CPU-intensive calculation in a more realistic problem.

These `openStore()`, `waitForCustomer()`, and `doneCutting()` actions by the barber and `getHairCut()` action by customers are straightforward manipulations of shared state while holding the lock and waiting as appropriate. Notice that in `waitForCustomer()` we use a variation of `wait()` that not only returns when signalled but also if some specified wallclock time is reached; we use this timeout to close the shop at the end of the day.

In `doneCutting()` we use `broadcast()` rather than `signal()` since several customers can be waiting, but only the customer with `custId == cutCount` can proceed.

```

// Wait until no active or waiting writers,
// then proceed.
void RWLock::startRead()
{
    lock.Acquire();
    waitingReaders++;
    while(readShouldWait()){
        readGo.Wait(&lock);
    }
    waitingReaders--;
    activeReaders++;
    lock.Release();
}

// Done reading. If no other active readers,
// a write may proceed.
void RWLock::doneRead()
{
    lock.Acquire();
    activeReaders--;
    if(activeReaders == 0
       && waitingWriters > 0){
        writeGo.Signal(&lock);
    }
    lock.Release();
}

// Read should wait if any active or waiting
// writer ("writers preferred" solution)
bool RWLock::readShouldWait()
{
    if(activeWriters > 0 || waitingWriters > 0){
        return true;
    }
    return false;
}

// Wait until no active readers or writers
// then proceed.
void RWLock::startWrite()
{
    lock.Acquire();
    waitingWriters++;
    while(writeShouldWait()){
        writeGo.Wait(&lock);
    }
    waitingWriters--;
    activeWriters++;
    lock.Release();
}

// Done writing. A waiting write or read
// may proceed.
void
RWLock::doneWrite()
{
    lock.Acquire();
    activeWriters--;
    assert(activeWriters == 0);
    if(waitingWriters > 0){
        writeGo.Signal(&lock);
    }
    else{
        readGo.Broadcast(&lock);
    }
    lock.Release();
}

// Write should wait if any active reader or writer
bool
RWLock::writeShouldWait()
{
    if(activeWriters > 0 || activeReaders > 0){
        return true;
    }
    return false;
}

```

Figure 5.15: RWLock.cc defines the implementation of our readers/writers solution

```
class BarberShop{
private:
    Lock lock;
    Cond wakeBarber;
    Cond nextCustomer;

    bool timeToClose;
    bool open;
    int arrivalCount;
    int cutCount;
    int fullCount;

public:
    BarberShop();
    ~BarberShop() {};
    void barberDay(); // Main loop for barber thread
    bool getHairCut(); // Called by customer thread
    void clockRingsClosingTime(); // Called by clock thread

private:
    void openStore();
    int waitForCustomer();
    void doneCutting();
    void printFinalStats();

    bool emptyAndOpen();
    bool stillNeedHaircut(int custId);
    bool waitingRoomFull();
};


```

Figure 5.16: BarberShop.h defines the interface and member variables for our Sleeping Barber solution.

```

void BarberShop::barberDay()
{
    // BarberDay is not an atomic action.
    // No lock. Only touch object's state
    // by calling methods that lock.
    int cust;
    printf("Opening for the day\n");
    openStore();
    while(1){
        cust = waitForCustomer();
        if(cust == NO_CUST_CLOSING_TIME){
            printf("Closing for the day\n");
            printFinalStats();
            return;
        }
        printf("Cut hair %d\n", cust);
        sthread_sleep(1, 0); // Simulate time to cut
        doneCutting();
    }
}

void BarberShop::openStore()
{
    lock.Acquire();
    open = true;
    lock.Release();
    return;
}

int BarberShop::waitForCustomer()
{
    int custId;
    lock.Acquire();
    while(emptyAndOpen()){
        wakeBarber.Wait(&lock);
    }
    if(timeToClose){
        open = false; // Stop new arrivals
    }
    if(arrivalCount > cutCount){
        custId = cutCount;
    }
    else{
        custId = NO_CUST_CLOSING_TIME;
    }
    lock.Release();
    return custId;
}

void BarberShop::doneCutting()
{
    lock.Acquire();
    cutCount++;
    nextCustomer.Broadcast(&lock);
    lock.Release();
    return;
}

void BarberShop::printFinalStats()
{
    lock.Acquire();
    printf("Stats: arrived=%d cut=%d full=%d\n",
          arrivalCount, cutCount, fullCount);
    assert(arrivalCount == cutCount);
    lock.Release();
}

```

```

bool BarberShop::getHairCut()
{
    int myNumber;
    bool ret;
    lock.Acquire();
    if(!open || waitingRoomFull()){
        ret = false;
    }
    else{
        // "Take a number" to ensure FIFO service
        myNumber = ++arrivalCount;
        wakeBarber.Signal(&lock);
        while(stillNeedHaircut(myNumber)){
            nextCustomer.Wait(&lock);
        }
        ret = true;
    }
    lock.Release();
    return ret;
}

void BarberShop::clockRingsClosingTime()
{
    lock.Acquire();
    timeToClose = true;
    wakeBarber.Signal(&lock);
    lock.Release();

    // Internal functions for checking status.
    // Always called with lock already held.
    bool BarberShop::emptyAndOpen()
    {
        if((timeToClose || (arrivalCount > cutCount))){
            return false;
        }
        else{
            return true;
        }
    }

    bool BarberShop::stillNeedHaircut(int custId)
    {
        // Ensure FIFO order by letting customers
        // leave in order they arrive
        if(custId > cutCount){
            return true;
        }
        else{
            return false;
        }
    }

    bool BarberShop::waitingRoomFull()
    {
        // +1 b/c barber chair
        if(arrivalCount - cutCount == NCHAIRS + 1){
            return true;
        }
        else{
            return false;
        }
    }
}

```

Figure 5.17: BarberShop.cc defines our Sleeping Barber solution

Exercises

9. In the readers-writers lock example for the function `RWLock::doneRead()`, why do we use `writeGo.Signal()` rather than `writeGo.Broadcast()`?
10. Show how to implement a semaphore by generalizing the the multi-processor lock implementation shown in Figure 5.12.
11. On page page 189, we sketched part of a solution to the Too Much Milk problem. To make the problem more interesting, we will also allow roommates to drink milk.

Implement in C++ or Java a `Kitchen` class with a `drinkMilkAndBuyIfNeeded()`. This method should randomly (with a 20% probability) change the value of `milk` from 1 to 0. Then, if the value just became 0, it should buy milk (incrementing `milk` back to 1). The method should return 1 if the roommate bought milk and 0 otherwise.

Your solution should use locks for synchronization and it should work for any number of roommates. Test your implementation by writing a program that repeatedly creates a `Kitchen` object and varying numbers of roommate threads; each roommate thread should call `drinkMilkAndBuyIfNeeded()` multiple times in a loop.

Hint: You will probably write a `main()` thread that creates a `Kitchen` object, creates multiple roommate threads, and then waits for all of the roommates to finish their loops. If you are writing in C++ with the Posix threads library, you can use `pthread_join()` to have one thread wait for another thread to finish. If you are writing in Java with the `java.lang.Thread` class, you can use the `join()` method.

12. For the solution to Too Much Milk suggested in the previous problem, each call to `drinkMilkAndBuyIfNeeded()` is atomic and holds the lock from the start to the end even if one roommate goes to the store. This solution is analogous to the roommate padlocking the Kitchen while going to the store, which seems a bit unrealistic.

Implement a better solution to `drinkMilkAndBuyIfNeeded()` using both locks and condition variables. Since a roommate now needs to release the lock to the kitchen while going to the store, you will no longer acquire the lock at the start of this function and release it at the end. Instead, this function will call two helper-functions, each of which acquires/releases the lock. E.g.,

```
int
Kitchen::drinkMilkAndBuyIfNeeded(){
    int iShouldBuy = waitThenDrink();
    if(iShouldBuy){
        buyMilk();
    }
}
```

In this function, `waitThenDrink()` should (if there is no milk) wait (using a condition variable) until there is milk, drink the milk, and if the milk is now gone, return a nonzero value to flag that the caller should buy milk. `BuyMilk()` should buy milk and then broadcast to let the waiting threads know that they can proceed.

Again, test your code with varying numbers of threads.

13. Before entering a *priority critical section*, a thread calls `PriorityLock::enter(priority)` and when the thread exits such a critical section it calls `PriorityLock::exit()`.
If several threads are waiting to enter a priority critical section the one with the numerically highest priority should be the next one allowed in. Implement `PriorityLock` using monitors (locks and condition variables) and following the multi-threaded programming standards defined for the class.
 - (a) Define the state and synchronization variables and describe the purpose of each.
 - (b) Implement `PriorityLock::enter(int priority)`
 - (c) Implement `PriorityLock::exit()`

5.7 Conclusions

Using well-structured shared objects to share state among threads makes reasoning about multithreaded programs vastly simpler than it would be if we tried to reason about the possible interleavings of individual loads and stores.

Furthermore, if we follow a systematic approach, it is not difficult to write code for shared objects that is easy for us to reason about and for others to read, understand, maintain, and change.

In short, this chapter defines a set of core skills that almost any programmer will use over and over again during the coming decade or longer.

That is not the whole story. As the next chapter will discuss, as systems grow to include many shared objects and threads, new challenges arise: synchronizing operations that span multiple shared objects, avoiding deadlocks in which a set of threads are all waiting for each other to do something, and maximizing performance when large numbers of threads are contending for a single object. Sadly, solutions to these problems are not as cut and dried.

Exercises

For convenience, the exercises from the body of the chapter are repeated here.

1. Show that solution 3 to the Too Much Milk problem is safe—that it guarantees that at most one roommate buys milk.
2. Precisely describe the set of possible outputs that could occur when the program shown in Figure 5.5 is run.
3. Suppose that a programmer mistakenly creates an automatic (aka local) variable v in one thread $t1$ and passes it to another thread $t2$. Is it possible for a write by $t1$ to some variable other than v will change the state of v as observed by $t2$? If so, explain how this can happen and give an example. If not, explain why not.
4. Suppose that a programmer mistakenly creates an automatic (aka local) variable v in one thread $t1$ and passes it to another thread $t2$. Is it possible for a write by $t2$ to v will cause $t2$ to execute the wrong code? If so, so, explain how. If not, explain why not.
5. Assuming Hansen semantics for condition variables, our implementation of the blocking bounded queue in Figure 5.10 does not guarantee freedom from starvation: if a continuous stream of threads makes `insert()` (or `remove()`) calls, it is possible for a waiting thread to wait forever. For example, a thread may call `insert()` and wait in the `while(isFull())` loop; then, every time another thread calls `remove()` and signals on the `itemRemoved` condition variable, a *different* thread might call `insert()`, see that the queue is not full, and insert an item before the waiting thread resumes. Then, when the waiting thread resumes, it will retest the `isFull()` predicate, see that the queue is full, and `wait()`.
Prove that under Hoare semantics and assuming that when a signal occurs, it is the longest-waiting thread that is resumed, our implementation of BBQ ensures freedom from starvation. That is, that if a thread waits in `insert()`, then it is guaranteed to proceed after a bounded number of `remove()` calls complete, and vice versa.
6. As noted in the previous problem, our implementation of the blocking bounded queue in Figure 5.10 does not guarantee freedom from starvation. Modify the code to ensure freedom from starvation so that if a thread waits in `insert()`, then it is guaranteed to proceed after a bounded number of `remove()` calls complete, and vice versa. **Note:** Your implementation must work under *Hansen/Mesa semantics* for condition variables.

7. Wikipedia provides an implementation of Peterson's algorithm to provide mutual exclusion using loads and stores at [http://en.wikipedia.org/wiki/Peterson's_algorithm](http://en.wikipedia.org/wiki/Peterson%27s_algorithm). Unfortunately, this code is not guaranteed to work with modern compilers or hardware. Update the code to include memory barriers where necessary. (Of course you could add a memory barrier before and after each instruction; your solution should instead add memory barriers only where necessary for correctness.)
8. Linux provides a `sys_futex()` system call to assist in implementing hybrid user-level/kernel-level locks and condition variables.

A call to `long sys_futex(void *addr1, FUTEX_WAIT, int val1, NULL, NULL, 0)` checks to see if the memory at address `addr1` has the same value as `val1`. If so, the calling thread is suspended. If not, the calling thread returns immediately with the error return value `EWOULDBLOCK`. In addition, the system call will return with the value `EINTR` if the thread receives a signal.

A call to `long sys_futex(void *addr1, FUTEX_WAKE, 1, NULL, NULL, 0)` causes one thread waiting on `addr1` to return.

Consider the following (too) simple implementation of a hybrid user-level/kernel-level lock.

```
class TooSimpleFutexLock{
    private:
        int val;

    public:
        TooSimpleMutex() : val(0) {} // Constructor

        void Acquire () {
            int c;
            while ((c = atomic_inc(&val)) != 0){ // atomic_inc returns *old* value
                futex_wait(&val, c + 1);
            }
        }

        void Release () {
            val = 0;
            futex_wake(&val, 1);
        }
};
```

There are three problems with this code.

- (a.) **Performance.** The goal of this code is to avoid making (expensive) system calls in the uncontested case when an `Acquire()` tries to acquire a free lock or a `Release()` call releases a lock with no other waiting threads. This code fails to meet this goal. Why?
- (b.) **Performance.** There is a subtle corner case when multiple threads try to acquire the lock at the same time that can show up as occasional slowdowns and bursts of CPU usage. What is the problem?

- (c.) **Correctness.** There is a corner case that can cause the mutual exclusion correctness condition to be violated, allowing two threads to both believe they hold the lock. What is the problem?
9. In the readers-writers lock example for the function `RWLock::doneRead()`, why do we use `writeGo.Signal()` rather than `writeGo.Broadcast()`?
 10. Show how to implement a semaphore by generalizing the the multi-processor lock implementation shown in Figure 5.12.
 11. On page page 189, we sketched part of a solution to the Too Much Milk problem. To make the problem more interesting, we will also allow roommates to drink milk.

Implement in C++ or Java a `Kitchen` class with a `drinkMilkAndBuyIfNeeded()`. This method should randomly (with a 20% probability) change the value of `milk` from 1 to 0. Then, if the value just became 0, it should buy milk (incrementing `milk` back to 1). The method should return 1 if the roommate bought milk and 0 otherwise.

Your solution should use locks for synchronization and it should work for any number of roommates. Test your implementation by writing a program that repeatedly creates a `Kitchen` object and varying numbers of roommate threads; each roommate thread should call `drinkMilkAndBuyIfNeeded()` multiple times in a loop.

Hint: You will probably write a `main()` thread that creates a `Kitchen` object, creates multiple roommate threads, and then waits for all of the roommates to finish their loops. If you are writing in C++ with the Posix threads library, you can use `pthread_join()` to have one thread wait for another thread to finish. If you are writing in Java with the `java.lang.Thread` class, you can use the `join()` method.

12. For the solution to Too Much Milk suggested in the previous problem, each call to `drinkMilkAndBuyIfNeeded()` is atomic and holds the lock from the start to the end even if one roommate goes to the store. This solution is analogous to the roommate padlocking the Kitchen while going to the store, which seems a bit unrealistic.

Implement a better solution to `drinkMilkAndBuyIfNeeded()` using both locks and condition variables. Since a roommate now needs to release the lock to the kitchen while going to the store, you will no longer acquire the lock at the start of this function and release it at the end. Instead, this function will call two helper-functions, each of which acquires/releases the lock. E.g.,

```
int
Kitchen::drinkMilkAndBuyIfNeeded(){
    int iShouldBuy = waitThenDrink();
```

```
    if(iShouldBuy){  
        buyMilk();  
    }  
}
```

In this function, `waitThenDrink()` should (if there is no milk) wait (using a condition variable) until there is milk, drink the milk, and if the milk is now gone, return a nonzero value to flag that the caller should buy milk. `BuyMilk()` should buy milk and then broadcast to let the waiting threads know that they can proceed.

Again, test your code with varying numbers of threads.

13. Before entering a *priority critical section*, a thread calls `PriorityLock::enter(priority)` and when the thread exits such a critical section it calls `PriorityLock::exit()`. If several threads are waiting to enter a priority critical section the one with the numerically highest priority should be the next one allowed in. Implement `PriorityLock` using monitors (locks and condition variables) and following the multi-threaded programming standards defined for the class.
 - (a) Define the state and synchronization variables and describe the purpose of each.
 - (b) Implement `PriorityLock::enter(int priority)`
 - (c) Implement `PriorityLock::exit()`

Chapter 6

Advanced Synchronization

Measure twice. Cut once.

– Carpenter’s motto

The biggest speedup your program will ever see is when it goes from “not working” to “working.” That’s infinite speedup.

– John Ousterhout

Premature optimization is the root of all evil (or at least most of it) in programming.

– Don Knuth

When two trains approach each other at a crossing, both shall come to a full stop and neither shall start up again until the other has gone.

– Kansas law, 1920s

The shared objects described in Chapter ?? provide a key building block for writing multi-threaded programs, but many such programs must address several additional issues.

The first set of issues arise because many programs comprise multiple shared objects, and we need to reason about the interactions among these pieces. Unfortunately, these interactions can break modularity; in some cases when one module calls another, you literally have to know about the modules' internal implementation details to make sure that both modules' synchronization mesh. There are two issues: safety and liveness.

1. **Safety: Multi-object synchronization.** For programs with multiple shared objects, we face a problem similar to what we faced when reasoning about atomic loads and stores: even if each individual operation on a shared object is atomic, we need to reason about interactions of sequences of operations across objects.
2. **Liveness: Deadlock.** One way to help reason about sequences of operations on multiple objects is to hold multiple locks. This approach raises the issue of deadlock where a set of threads get permanently stuck waiting for each other in a cycle.

The bad news here is that there is no cookbook recipe that always works for dealing with these challenges. In particular, current techniques for addressing these problems have two basic limitations. First, there are engineering trade-offs among them. Some solutions are general but complex or expensive; others are simple but slow; and still others are simple and cheap but not general. Second, many of the solutions are inherently *non-modular*. They require reasoning about the global structure of the system and internal implementation details of modules to understand or restrict how different modules can interact.

In addition to discussing the challenges that arrive when dealing with multiple objects and multiple locks, this chapter discusses one other issue: how to construct shared objects that can be accessed without locks. We emphasize that this is an advanced topic that should only be considered by programmers that have mastered multi-threaded programming. The vast majority of the time, the simple shared objects described in the last chapter will be all that is needed in a multi-threaded program.

3. **Synchronization with reduced locking.** In the part of this chapter, we briefly discuss two techniques for synchronizing access to shared state without locking: *read-copy-update (RCU)* and *lock-free/wait-free data structures*.

6.1 Multi-object synchronization

Having multiple shared objects in a program raises challenges to reasoning about interactions across objects. For example, consider a system storing a bank’s accounts. A reasonable design choice might be for each customer’s account to be a shared object with a lock (either a mutual exclusion lock or a readers/writers lock as described in the previous chapter.) Consider, however, transferring \$100 from account *A* to account *B* as follows:

```
A->subtract(100);
B->add(100);
```

Although each individual action is atomic, the sequence of actions is not. As a result, there may be a time where, say, *A* tells *B* that the money has been sent and *B* gets mad because the money does not appear in *B*’s account.

Similarly, consider a bank manager running a program to answer a question: “How much money does the bank have?” If the program simply reads from each account, the calculation may exclude or double-count money “in flight” between accounts such as in the transfer from *A* to *B*.

These examples illustrate a general problem that arises whenever a program contains multiple shared objects accessed by threads. Even if each object has a lock and guarantees that its methods operate atomically, *sequences* of operations by different threads across different objects can be interleaved. For example, you would face the same issues if you tried to solve Too Much Milk with a **Note** object that has 2 methods, `readNote` and `writeNote`, and a **Fridge** object that has 2 methods, `checkForMilk` and `addMilk`.

One big lock v. fine-grained locking. The simplest solution is to include all of a program’s data structures in a single shared object with a single lock. Then, all threads operate by calling that object’s methods, each of which can operate while holding the object’s lock. This approach can yield acceptable performance for some applications, especially if the code is structured so that threads do not hold the lock when they perform high-latency I/O operations.

However, for other applications, a single global lock may restrict parallelism too much. In these cases, different data structures may each have their own lock.

By the same token, although the previous chapter focused on the simple case of a shared object with a single lock protecting all of the object’s state, *fine-grained locking*—partitioning an object’s state into different subsets protected by different locks—is sometimes warranted.

Definition: **fine-grained locking**

Example: Hash table with fine grained locking. A hash table provides `put(key, value)`, `value = get(key)`, and `value = remove(key)` methods. A simple coarse-grained locking implementation would use a single lock that is acquired and released at the start and end of each of these methods. If

serializing all requests limits performance, a fine-grained alternative is to have one lock per hash bucket and to acquire the lock for bucket b before accessing any record that hashes to bucket b .

There is no fundamental difference between a program with multiple shared objects, each with its own lock, and a shared object that uses fine grained locking and that has multiple locks covering different subsets of its data structures. All of our discussions about issues that arise when accessing multiple shared objects in a program also apply to fine-grained locking within an object.

Complexity v. performance. Beware of premature optimization: dividing an object's state into different pieces protected by different locks can significantly increase the object's complexity and does not always significantly improve performance.

Example: Resizable hash table. Suppose we want to implement a hash table whose number of hash buckets grows as the number of objects it stores increases. If we have a single lock, this is easy to do. But, what if we use fine-grained locking? Then the design becomes more complex because we have some operations like `put()` and `get()` that operate on one bucket and other operations like `resize()` that operates across multiple buckets.

One solution is to have a readers-writers lock on the overall structure of the table (e.g., the number of buckets and the array of buckets) and a mutual exclusion locks on each bucket. Then, `put()` and `get()` acquire the table's readers-writers lock in read mode and also acquire the relevant bucket's mutual exclusion lock, and `resize()` acquires the readers-writers lock in write mode.

A second solution is to have one mutual exclusion lock for each bucket, for `get()` and `put()` to acquire the relevant bucket's mutual exclusion lock, and for `resize()` to iterate through the buckets, acquiring all of the buckets' locks.

A third solution is to divide the hash key space into r regions, to have a mutual exclusion lock for each region, and to allow each region to be resized independently when that region becomes heavily loaded. Then, `get()`, `put()`, and `resizeRegion()` each acquire the relevant region's mutual exclusion lock.

Which is solution is best? It is not obvious. The first solution is simple and appears to allow good concurrency, but acquiring the readers-writers lock even in read mode often involves writing a cache line that will be shared by many processors, so it may have poor cache performance. The second solution makes `resize()` expensive, but if `resize()` is a rare operation, that may be OK. The third solution could balance concurrency for `get()`/`put()` against the cost of `resize()`, but it is more complex and may require tuning the number of groups to get good performance. And, these trade-offs may change as the implementation becomes more complex; for example to trigger `resize()` at appropriate times, we probably need to maintain an additional `nObjects` count of the number of objects currently stored in the hash table, so whatever locking approach we use would need to be extended to cover this information.

Often, the best practice is to start simple, often with a single-lock per shared object. If the objects' interfaces are well designed, then refactoring their implementations to increase concurrency and performance can be done once the system is built and performance measurements have identified the bottlenecks. “It is easier to go from a working system to a working, fast system than to go from a fast system to a fast, working system.”

Example: Linux evolution. The first versions of Linux ran only on uniprocessor machines. To allow Linux to run on multiprocessor machines, version 2.0 introduced the Big Kernel Lock (BKL)—a single lock that protected all of the kernel's shared data structures. The BKL allowed the kernel to function on multiprocessor machines, but scalability and performance were limited. So, over time, different subsystems and different data structures got their own locks, allowing them to be accessed without holding the BKL. By version 2.6, Linux has been highly optimized to run well on multiprocessor machines—Linux now has thousands of different locks and researchers have demonstrated scalability for a range of benchmarks on a 48 processor machine. Still, the BKL remains in use in a few—mostly less performance-critical—parts of the Linux kernel like the `reboot()` system call, some older file systems, and some device drivers.

6.1.1 Solutions and design patterns

As noted above, there are no completely general solutions for how to structure multi-object and multi-lock programs. However, there are approaches and design patterns that work well in practice. We discuss four examples here and more examples after we have addressed deadlock, which also affects techniques for writing multi-object, multi-lock programs.

Careful class design

It can be easier to reason about sequences of operations on objects than to reason about sequences of atomic reads and writes of memory because we often have control over the interface to those objects. Careful class and interface design can make it feasible to reason about the overall program. This need for careful design includes the design of individual objects (e.g., specifying clean interfaces that expose the right abstractions). It also includes the architecture of how those objects interact (e.g., structuring a system architecture in well-defined layers.)

Example: Too much milk. For example, as just noted, it would be difficult to solve Too Much Milk with a `Note` object and `Fridge` object with interfaces `Note::readNote()`, `Note::writeNote()`, `Fridge::checkForMilk()`, and `Fridge::addMilk()`. On the other hand, if we refactor the objects so that we have `Fridge::checkforMilkAndSetNoteIfNeeded()` and `Fridge::addMilk()`, the problem is straightforward.

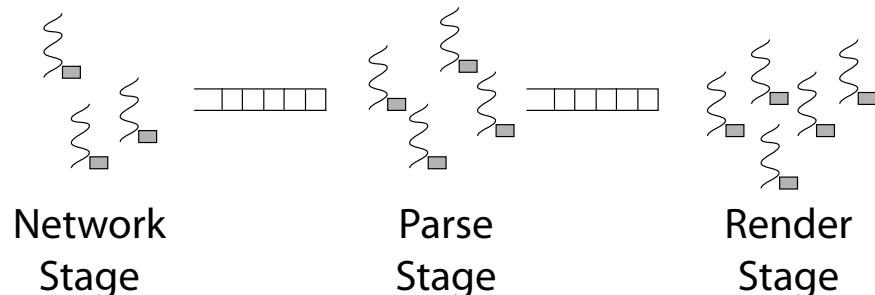


Figure 6.1: A multi-stage server based on the ownership pattern. In the first stage, one thread exclusively owns each network connection. In later stages, one thread is processing a given object at a time.

This advice is admittedly rather generic and perhaps obvious: of course one should strive for elegant designs for both single- and multi-threaded code. Nonetheless, we emphasize that the choices you make for your interfaces, abstractions, and software architecture can dramatically affect the complexity or feasibility of your designs.

Ownership pattern

One common synchronization technique in large, multi-threaded programs is an **ownership design pattern** in which a thread removes an object from a container and then may access the object without holding a lock because the program structure guarantees that at most one thread owns an object at a time.

Example: Work queue. A single web page can contain multiple objects including html frames, style sheets, and images. Consider a multi-threaded web browser whose processing is divided into three stages: receiving an object via the network, parsing the object, and rendering the object (see Figure 6.1.) For the first stage, we have one thread per network connection, and for the other stages we have several worker threads, each of which processes one object at a time.

The work queues between stages coordinate object ownership. Objects in the queues are not being accessed by any thread. When a worker thread in the *parse* stage removes an object from the stage's work queue, it owns the object and has exclusive access to it. When it is done parsing the object, it puts the parsed object into the second queue and stops accessing it. A worker thread from the *render* stage then removes it from the second queue, gaining exclusive access to it to render it to the screen.

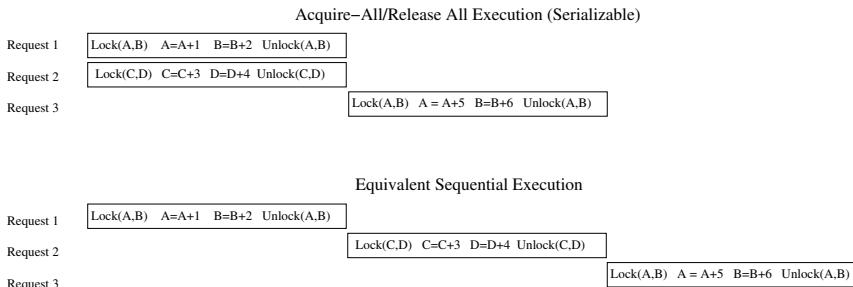


Figure 6.2: Locking multiple objects using an acquire-all/release-all pattern results in a serializable execution that is equivalent to an execution where requests are executed sequentially in some order.

Acquire-all/release-all

If a system that processes requests has multiple locks, one way to manage these locks is the *acquire-all/release-all pattern*. To process a request in this pattern, a thread first acquires all of the locks that will be needed at any point during request processing, then the thread processes the request, finally the thread releases all of the locks.

Definition:
acquire-all/release-all pattern

This approach can allow significant concurrency. If requests touch nonoverlapping subsets of state protected by different locks, then they can proceed in parallel.

This approach can be easy to reason about because it enforces *serializability* across requests—the result of any execution of the program is equivalent to an execution in which requests are processed one at a time in some sequential order. As Figure 6.2 illustrates, requests that access nonoverlapping data can proceed in parallel. The result is the same as it would have been if, instead, the system first executed one of the requests and then the other. On the other hand, if two requests touch any of the same data, then one will be processed entirely before the other’s processing begins. Ensuring serializability thus allows one to reason about multi-step tasks *as if* each task executed alone.

Definition: **serializability**

Example: Hash table. Consider a hash table with one lock per hash bucket and that supports a `changeKey(k1, k2)` operation that changes the object that initially has key `k1` to have key `k2` instead. This function could be implemented to acquire `k1` and `k2`'s locks, remove the object using key `k1`, update the object's key, insert the object using key `k2`, and then release `k1` and `k2`'s locks.

One challenge to using this approach is knowing exactly what locks will be needed by a request before beginning to process it. One solution is to conservatively acquire more locks than needed (e.g., acquire any locks that *may* be needed by a particular request), but this approach can reduce concurrency.

Another challenge is that locks may be held for longer than needed. This aspect of the approach can also reduce concurrency.

Two-phase locking refines the acquire-all/release-all pattern to address these two challenges.

Two-phase locking

Definition: **two phase locking** In *two phase locking* a multi-step task is divided into two phases. During the *expanding* phase, locks may be acquired but not released. Then, in the *contracting* phase, locks may be released but not acquired.

For some programs, this approach can support more concurrency than the acquire-all/release-all pattern. Because locks can be acquired during the expanding phase, two-phase locking does not require deciding what locks to grab *a priori*, so programs may be able to avoid acquiring locks they do not end up needing, and they may not need to hold some locks for as long.

Two phase locking pattern facilitates reasoning about programs because it also ensures that all executions are serializable. To see this, notice that if two requests access overlapping data, then one of them will lock all of the overlapping data before the other one begins its access to that data; then, once the thread processing the first request begins releasing locks and the thread processing the second request begins acquiring locks, the second request only modifies data that the first request will not access again. The execution thus appears as it would have if the first request finished accessing all of the overlapping data before the second request accesses any of the overlapping data, which, in turn, appears as it would have if the first request finished executing before the second request began executing.

Example: Hash table. A `changeKey(k1, k2)` function for a hash table with per-bucket locks could be implemented to acquire `k1`'s lock, remove the object using key `k1`, update the object's key, acquire `k2`'s lock, release `k1`'s lock, insert the object using key `k2`, and release `k2`'s lock.

Staged architectures

Definition: **staged architecture** One common pattern is the *staged architecture* pattern, illustrated in Figure 6.3. A staged architecture divides a system into multiple subsystems called stages, where each stage includes some state private to the stage and a set of one or more worker threads that operate on that state. Different stages communicate by sending messages to each other via shared producer-consumer queues, and each worker thread repeatedly pulls the next message from a stage's incoming queue and then processes it, possibly producing one or more messages for other stages' queues.

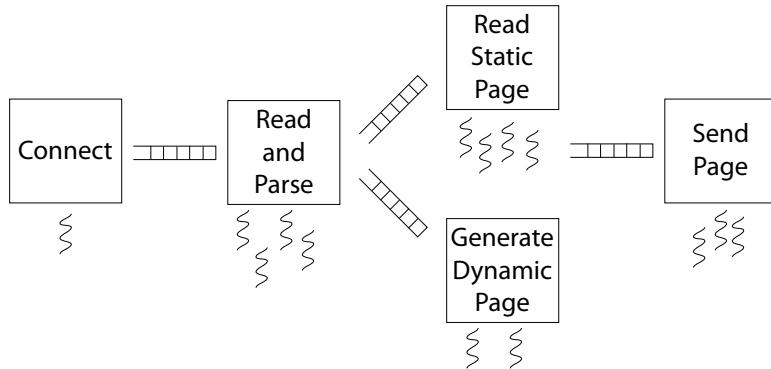


Figure 6.3: A staged architecture for a simple web server.

Example: Simple web server. Figure 6.3 shows a staged architecture for a simple web server that has a first *connect* stage that uses 1 thread to set up network connections and that passes each connection to a second *read and parse* stage.

The *read and parse* stage has several threads, each of which repeatedly gets a connection from the incoming queue, reads a request from the connection, and parses the request to determine what web page is being requested.

If the request is for a static web page (e.g., an HTML file), the *read and parse* stage passes the request and connection to the *read static page* stage, where one of the stage's threads reads the specified page from disk. Otherwise, the *read and parse* stage passes the request and connection to the *generate dynamic page* stage, where one of the stage's threads runs a program that dynamically generates a page in response to the request.

Once the page has been fetched or generated, the page and connection are passed to the *send page* stage, where one of the threads transmits the page over the connection.

The key property of a staged architecture is that the state of each stage is private to that stage. This property improves modularity, making it easier to reason about each stage individually and about interactions across stages.

As an example of the modularity benefits, consider implementing a system where different stages are produced by different teams or even different companies. Each stage can be designed and tested almost independently, and the system is likely to work as expected when the stages are brought together. For example, it is common practice for a web site to use a web server from one company and a database from another company and for the two to communicate via messages.

Another benefit of this approach for some applications is improved cache locality. If a thread on a processor is operating on a subset of the system's state, it may have a better cache hit rate than a thread that must access state

from all stages. On the other hand, for some workloads, passing a request from stage to stage will hurt cache hit rates compared to doing all of the processing for a request on one processor.

Also note that for good performance, the processing in each stage must be large enough to amortize the cost of sending and receiving messages.

The special case when there is exactly one thread per stage is called *event*

Definition: event processing *processing*. A special property of event processing architectures is that there is no concurrency within a stage, so no locking is required and each message is processed atomically with respect to that stage's state.

Overload. One challenge with staged architectures is dealing with overload. The throughput of the system will be limited by that of the slowest stage. If the system is overloaded, the slowest stage will fall behind and the queue before it will grow. Depending on the system's implementation, two bad things can happen. First, the queue can grow indefinitely, consuming more and more memory until the system runs out of memory. Second, if the queue is limited to a finite size, once that size is reached, earlier stages must either discard messages they want to send to the overloaded stage or they must block until the queue has room. Notice that if they block, then the backpressure will limit earlier stages' throughput to that of the bottleneck stage, and their queues may begin to grow.

One solution is to dynamically vary the number of threads per stage. If a stage's incoming queue is growing, shift processing resources to it by stopping one of the threads for a stage with a short queue start a new thread for the stage that is falling behind.

6.2 Deadlock

A challenge to constructing programs that include multiple shared objects is **deadlock**. *Deadlock* is a cycle of waiting among a set of threads where each thread is waiting for some other thread in the cycle to take some action.

Definition: mutually recursive locking Figure 6.4 shows two examples of deadlock. In *mutually recursive locking*, code in each of two shared objects $s1$ and $s2$ holds a lock while calling into a method in the other shared object that uses that object's lock. Then, threads 1 and 2 can deadlock if thread 1 calls a method in $s1$ that holds the lock and tries to call a method in $s2$ that needs a lock while thread 1 calls a method in $s2$ that holds $s2$'s lock and that tries to call a method in $s1$ that needs $s1$'s lock.

Definition: nested waiting In *nested waiting*, code in one shared object $s1$ calls a method of another shared object $s2$, which waits on a condition variable. The condition variable's `wait()` method releases $s2$'s lock but not $s1$'s, so the thread that would have done a signal in $s2$ may stuck waiting for $s1$'s lock.

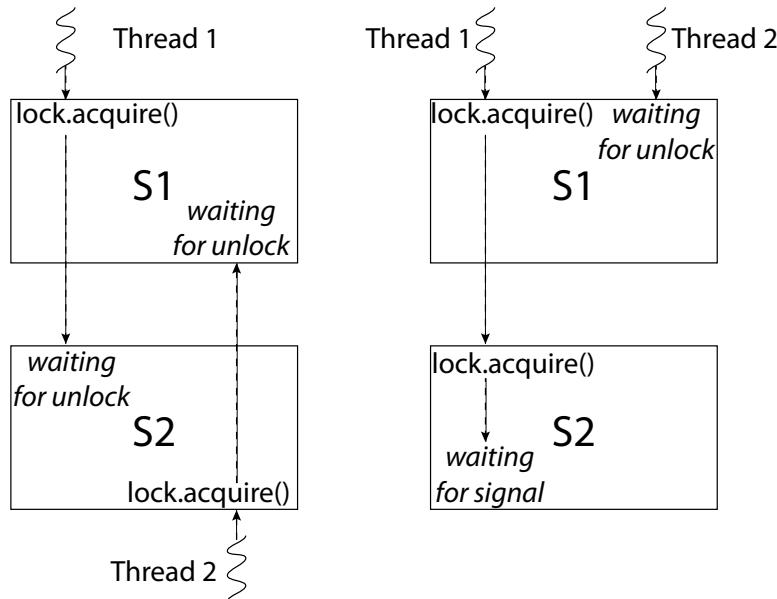


Figure 6.4: Two examples of deadlock: *mutually recursive locking* (left) and *nested waiting* (right).

Example: The Dining Philosophers.

The Dining Philosophers problem is a classic synchronization problem that illustrates the challenge of deadlock. There is a round table with n plates and n chopsticks arranged as illustrated in Figure 6.5. A philosopher sitting at each plate requires two chopsticks to eat. Suppose that each philosopher proceeds by grabbing the chopstick on the left, grabbing the chopstick on the right, eating, and then replacing both chopsticks. If philosophers follow this approach they can unfortunately enter a deadlock: each philosopher can grab the chopstick on the left but then be stuck waiting for the philosopher on the right to release the chopstick she holds.

Deadlock v. starvation. Deadlock and starvation are both liveness concerns. In *starvation*, some thread fails to make progress for an indefinite period of time. Deadlock is a form of starvation but with the stronger condition that a group of threads form a cycle where none of the threads make progress because each thread is waiting for some other thread in the cycle. Thus, deadlock implies starvation (literally, for the dining philosophers), but starvation does not imply deadlock.

Definition: **starvation**

For example, recall the readers/writers example discussed in Section 5.6.8. If instead of implementing a writers-preferred solution we implement a readers-preferred solution where a reader only waits if a writer is currently active, then

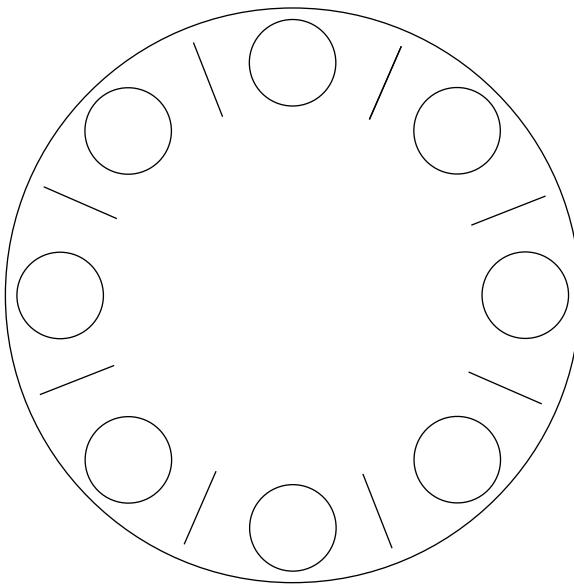


Figure 6.5: In this example of the dining philosophers problem, there are 8 philosophers, 8 plates, and 8 chopsticks.

writers could starve if the workload includes a large number of readers. Note that such starvation would not be deadlock—the writers are waiting on the readers, but the readers are not waiting on the writers.

Nondeterminism. Just because a system might suffer a deadlock or might starve a thread does not mean that it always will. A system is *subject to starvation* if it is possible for a thread to starve in some circumstances. A system is *subject to deadlock* if it is possible for a group of threads to enter deadlock in some circumstances. Here, the circumstances that affect whether deadlock or starvation occurs may include a broad range of factors such as the choices made by the scheduler, the number of threads running, the workload or sequence of requests processed by the system, which threads win races to acquire locks, and which threads are enabled in what order when signals or broadcasts occur.

A system that is subject to starvation or deadlock may be live in many or most runs and only starve or deadlock for particular workloads or “unlucky” interleavings. For example, in the *mutually recursive locking* example in Figure 6.4, the deadlock only occurs if the threads call the indicated functions at about the same time, and for the Dining Philosophers problem, philosophers may succeed in eating for a long time before hitting the unlucky sequence of events that causes them to deadlock. Similarly, in the readers/writers example, the readers-preferred solution will allow some writes to complete as long as the rate of reads stays below some threshold.

Since testing may not uncover deadlock problems, it is important to construct systems that are deadlock-free by design.

6.2.1 Necessary conditions for deadlock

There are four necessary conditions for deadlock to occur. Knowing these conditions is useful for designing solutions to deadlock: if you can prevent any one of these conditions, then you can eliminate the possibility of deadlock.

1. **Bounded resources.** There are a finite number of threads that can simultaneously use a resource.
2. **No preemption.** Once a thread acquires a resource, the ownership of the resource cannot be revoked until the thread acts to release it.
3. **Wait while holding.** A thread holds one resource while waiting for another. This condition is sometimes called *multiple independent requests* because it occurs when a thread first acquires one resource and then attempts to acquire another resource.
4. **Circular waiting.** There is a set of waiting threads such that each thread is waiting for a resource held by another.

Example: Dining Philosophers. To illustrate the circular waiting condition, Figure 6.6 maps the state of a deadlocked Dining Philosophers implementation to an abstract graph that shows which resources are *owned by* which threads and which threads *wait for* which resources. In this type of graph, if there is one instance of each type of resource (e.g., a particular chopstick), then a cycle implies deadlock (assuming the system does not allow preemption.)

These four conditions are necessary but not sufficient for deadlock. If there are multiple instances of a type of resource, then there can be a cycle of waiting without deadlock because a thread not in the cycle may return resources to the pool.

Example: Dining Philosophers. Suppose we have a set of 5 philosophers at a table with 5 chopsticks but that the chopsticks are placed in a tray at the center of the table when they are not in use. We could be in the state illustrated in Figure 6.7 where philosopher 1 has two chopsticks, philosophers 2, 3, and 4 each have one chopstick and is waiting for another chopstick, and philosopher 5 has no chopsticks. In this state we have bounded resources (5 chopsticks), no preemption (we cannot forcibly remove a chopstick from a hungry philosopher's hand), wait while holding (philosophers 2, 3 and 4 are holding a chopstick while waiting for another), and circular waiting (each of philosophers 2, 3, and 4 are waiting for a resource held by another of them.) However, we do not have deadlock; eventually thread 1 will release its two

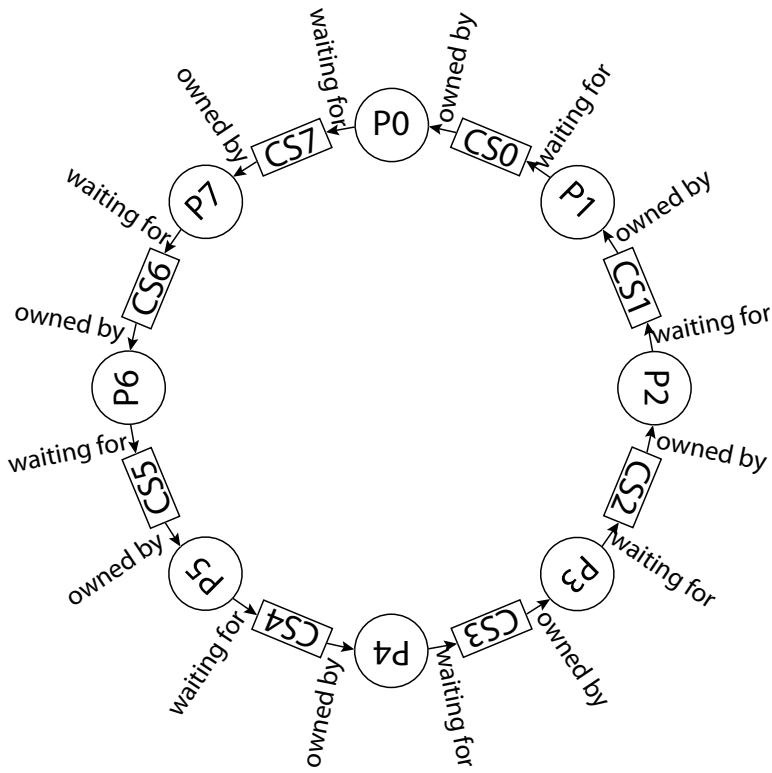


Figure 6.6: Graph representation of the state of a deadlocked Dining Philosophers system. Circles represent threads, boxes represent resources, an arrow from a box/resource to a circle/thread represents an *owned by* relationship and an arrow from a circle/thread to a box/resource represents a *waiting for* relationship.

chopsticks, which may, for example, allow threads 2 and 3 to eat and release their chopsticks which would allow threads 4 and 5 to eat.

Although the system shown in Figure 6.7 is not currently deadlocked, it is still subject to deadlock. For example, if philosopher 1 returns two chopsticks, philosopher 5 grabs one, and philosopher 1 grabs the other, then the system would deadlock.

6.2.2 Preventing deadlock

Preventing deadlock can be challenging. For example, consider a system with 3 resources—A, B, and C—and two threads that access them. Thread 1 acquires A then C then B and thread 2 acquires B then C then A. The following sequence can lead to deadlock:

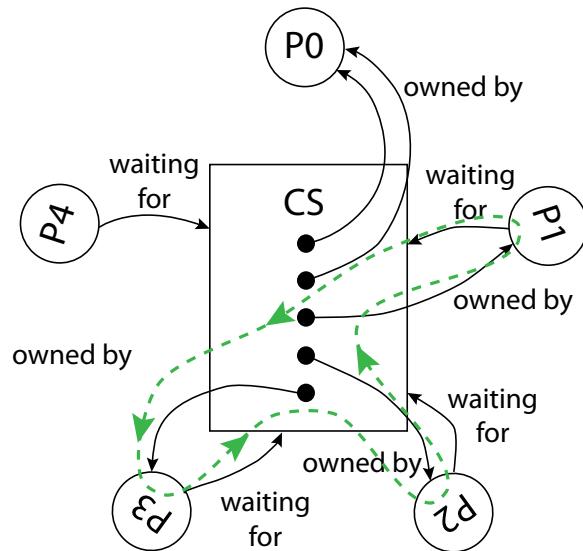


Figure 6.7: Graph representation of the state of a Dining Philosophers system that includes a cycle among waiting threads and resources but that is not deadlocked. Circles represent threads, boxes represent resources, dots within a box represent multiple instances of a resource, an arrow from a dot/resource instance to a circle/thread represents an *owned by* relationship and an arrow from a circle/thread to a box/resource represents a *waiting for* relationship.

	Thread 1	Thread 2
1	Acquire A	
2		Acquire B
3	Acquire C	
4		Wait for C
5	Wait for B	

How could we avoid this deadlock? The deadlock's circular waiting occurs when we reach step 5, but our fate was sealed much earlier. In particular, once we complete step 2 and thread 2 acquires B, deadlock is inevitable. In this case to prevent the deadlock we have to be smart enough to see the deadlock that will occur at step 5 much earlier; once step 1 completes and thread 1 acquires A, we cannot allow thread 2 to complete step 2 and acquire B or deadlock will follow.

This example illustrates that given an arbitrary program, preventing deadlock can be challenging. Deadlock prevention solutions, therefore, often restrict or take advantage of the structure of a program to avoid such complexity. As a result, no single solution is best in all cases.

Section 6.2.1 listed four necessary conditions for deadlock. These conditions are useful because they suggest approaches for preventing deadlock: if a system

is structured to prevent at least one of the conditions, then the system can not deadlock. Considering these conditions in the context of a given system often points to a viable deadlock prevention strategy. Below, we discuss some commonly-used approaches.

Bounded resources: Provide sufficient resources. One way to ensure deadlock freedom is to have sufficient resources to satisfy all threads' demands. For example, suppose an operating system allows a maximum of 10 open files per process, 10 processes, and 50 open files total; this system could deadlock for some workloads. On the other hand, if this system increases the global limit to 100 open files, then the open files resource cannot cause a deadlock.

No preemption: Preempt resources. Another technique is to allow the runtime system to forcibly reclaim resources held by a thread. For example, an operating system can preempt a page of memory from a running process by copying it to disk in order to prevent applications from deadlocking as they acquire memory pages.

No wait while holding: Abort request. Programs can choose not to wait for resources and abort a request that cannot immediately get all of the resources it needs. Although this approach sounds extreme, it can provide acceptable performance and can be much simpler than other alternatives for engineering deadlock out of a system.

For example, in old-style circuit-switched telephone networks, a call would have to reserve a circuit at a series of switches along its path. If the connection setup fails to find a free circuit at any hop, rather than wait for a circuit at the next hop to become free, it cancels the connection attempt, giving the user an error message (“All circuits are busy. Please try again later.”)

Similarly, when a router in the modern Internet is overloaded and runs out of packet buffers, it drops incoming packets. An alternative would be for each router to wait to send a packet until the next router has a buffer for it, but such an approach could deadlock.

No wait while holding: Atomically acquire all resources. Rather than acquiring resources in a sequence of steps, programs can be structured so that threads wait until all required resources are available and then acquire them all atomically. For example, a dining philosopher might wait until the two neighboring chopsticks are available and then simultaneously grab them both.

A thread may not know exactly which resources it will need to complete its work, but it can still acquire all resources that it *might* need when it begins work. For example, in an operating system for mobile phones where memory is constrained and where memory cannot be preempted by copying it to disk,

rather than having applications request additional memory as needed, we might instead have each application state its maximum memory needs and allocate that much memory when each application starts. Disadvantages of this approach include the challenge of having applications accurately estimate their worst-case needs and the cost of allocating significantly more resources than may be necessary in the common case.

No wait while holding: Release lock when calling out of module. If we have a series of nested modules, each of which has a lock, then waiting on a condition variable in an inner module can lead to a nested waiting deadlock. One solution is to restructure a module's code so that no locks are held when calling other modules. For example, we can change the code on the left to the code on the right:

```
Module::foo(){
    lock.acquire();
    doSomeStuff();
    otherModule->bar();
    doOtherStuff();
    lock.release();
}
Module::doSomeStuff(){
    x = x+1;
}
Module::doOtherStuff(){
    y = y-2;
}

Module::foo(){
    doSomeStuff();
    otherModule->bar();
    doOtherStuff();
}
Module::doSomeStuff(){
    lock.acquire();
    x = x+1;
    lock.release();
}
Module::doOtherStuff(){
    lock.acquire();
    y = y-2;
    lock.release();
}
```

Circular waiting: Lock ordering. An approach used in many systems is to identify an ordering among locks and to forbid acquiring a lock if any higher-ordered lock is already held.

For example, we can eliminate deadlock among the dining philosophers if—instead of always grabbing the chopstick on the left and then the one on the right, the philosophers number the chopsticks from 1 to n and always grab the lower-numbered chopstick before the higher-numbered one.

Similarly, for our hash table with per-bucket locks that supports a `changeKeys(k1, k2)` operation, we can avoid deadlock by always grabbing the lock for the lower-numbered bucket before the one for the higher-numbered bucket.

6.2.3 The Banker's Algorithm for avoiding deadlock

The Banker's Algorithm is another deadlock prevention approach. It is more complex to describe than the ones above, and systems seldom use it in its full generality. Nonetheless, we include this discussion both because simplified

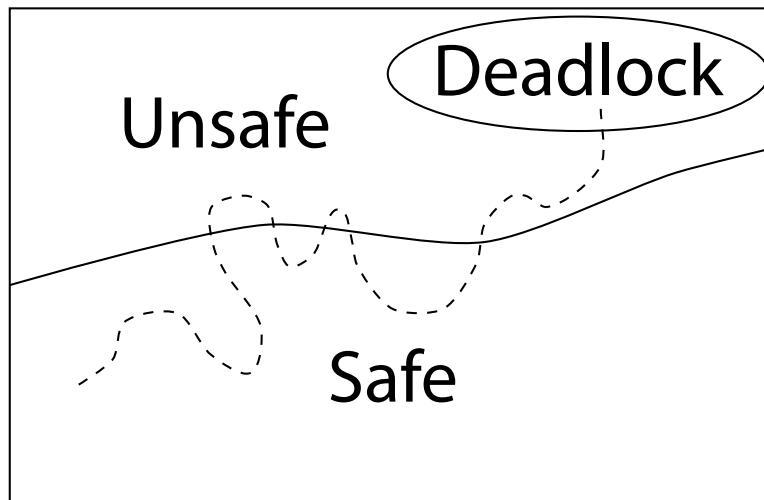


Figure 6.8: A process can be in a *safe*, *unsafe*, or *deadlocked* state. The dashed line illustrates a sequence of states visited by a thread—some are safe, some are unsafe, and the final state is a deadlock.

versions of the algorithm can be useful and because it sheds light on some underlying principles for understanding deadlocks such as the distinction between *safe* and *unsafe* states and how the occurrence of deadlocks often depends on a system’s workload and sequence of operations.

Dijkstra defined the *Banker’s Algorithm* as an approach that improves upon the *atomically acquire all resources* approach. A thread still *states* its maximum resource requirements when it begins a task, but it then acquires and releases those resources incrementally as the task runs. The runtime system *delays* granting some requests in a way that ensures the system never deadlocks.

The key idea behind the algorithm is that just because a system is capable of deadlock doesn’t mean that it must always deadlock: for some interleavings of requests it will deadlock, but for others it won’t. By delaying when some resource requests are processed, a system can avoid interleavings that could lead to deadlock.

A deadlock-prone system can be in one of three states: a *safe state*, an *unsafe state*, and a *deadlocked state* (see Figure 6.8.)

Definition: **safe state**

- In a *safe state*, for any possible sequence of resource requests, there exists at least one *safe sequence* of processing the requests that eventually succeeds in granting all pending and future requests.

Definition: **unsafe state**

- In an *unsafe state*, there exists at least one sequence of pending and future resource requests that leads to deadlock no matter what processing order is tried.

- In a *deadlocked state*, the system has at least one deadlock.

Definition: **deadlocked state**

So, as long as the system is in a safe state, it has control of its own destiny: for any workload, it can avoid deadlock by delaying the processing of some requests. In particular, the Banker's Algorithm will delay any request that would take it from a safe to an unsafe state because once the system enters an unsafe state, it may not be able to avoid deadlock.

Notice that an unsafe state does not always lead to deadlock. A system in an unsafe state may remain in an unsafe state or return to a safe state, depending on the specific interleaving of resource requests and completions. However, as long as the system is in an unsafe state, a bad workload or unlucky scheduling of requests can force it to deadlock.

The Banker's Algorithm is an approach to keeping a system in a safe state. The algorithm is based on a loose analogy with a small town banker who has a maximum total amount that can be loaned at one time `total` and a set of businesses that each have a credit line `max[i]` for business `i`. A business will borrow and pay back amounts of money as various projects are started and finished, so that the business `i` will always have an outstanding loan amount between 0 and `max[i]`. For each business with a credit line, if all of the business's requests are granted, the business will eventually reach a state where all current projects are finished and the loan balance returns to 0.

A conservative banker might only issue credit lines so that the sum of the credit lines is at most `total`; this approach is analogous to the *atomically acquire all resources* approach or the *provide sufficient resources* approach, and it trivially guarantees that the system remains in a safe state and that eventually all businesses with credit lines will complete their projects.

However, a more aggressive banker can issue more credit as long as the bank can live up to its commitment to each business—to provide a loan of `max[i]` if business `i` requests it. The key observation is that the bank can *delay* requests to increase a loan amount within a businesses existing credit line. For example, the bank might lose the paperwork for a few hours, days, or weeks.

By delaying loan requests, the bank can remain in a safe state—a state for which there exists at least one series of loan fulfillments by which every business `i` can eventually receive its maximal loan `max[i]`, complete its projects, and pay back all of its loan. The bank can then use that repaid money to grant pending loans to other businesses.

Figure 6.10 shows pseudo-code for a version of the banker's algorithm that manages a set of `r` resources for a set of `t` threads. For simplicity of discussion, threads request each unit of resource separately, but the algorithm can be extended to allow multiple resources to be requested at the same time.

The high-level idea is simple: when a request arrives, wait until it is safe to grant the request before granting it. As Figure 6.9 shows, we can realize this

```

class ResourceMgr{
private:
    Lock lock;
    Cond cv;
    int r; // Number of resources
    int t; // Number of threads
    int avail[]; // avail[i] is number of instances of resource i available
    int max[][]; // max[i][j] is max of resource i needed by thread j
    int alloc[][]; // alloc[i][j] is current allocation of resource i
                   // to thread j
}
...

```

Figure 6.9: State maintained by banker algorithm's resource manager. Resource manager code is in Figures 6.10 and 6.11.

```

// Invariant: the system is in a safe state
//
ResourceMgr::Request(int resourceId, int threadID){
    lock.Acquire();
    assert(isSafe());
    while(!wouldBeSafe(resourceID, threadID)){
        cv.Wait(&lock);
    }
    alloc[resourceID][threadID]++;
    avail[resourceID]--;
    assert(isSafe());
    lock.Release();
}

```

Figure 6.10: High level pseudo-code for banker's algorithm. The state maintained by the algorithm is defined in Figure 6.9 and the method `isSafe()` is defined in Figure 6.11.

high-level approach by tracking the current allocation of each resource to each thread, the maximum allocation possible for each thread, and the current set of available, unallocated resources.

Figure 6.11 shows how we can test whether a state is safe. Recall that a state is safe if there is some sequence of thread executions that will allow each thread to gain its maximum resource need, finish its work, and release its resources. So, we first see if the currently free resources would allow any thread to finish; if so, then the resources held by that thread would eventually be released to the system. Next, we see if the currently free resources plus the resources held by the thread identified in the first step would allow any other thread to finish; if so, both the first and second threads resources would eventually be released to the system. We continue this process until we have identified all threads that can be guaranteed to finish. If that set includes all of the threads, the state is safe.

Example: Page allocation with the Bankers Algorithm. Suppose we have a system with 8 pages of memory and three processes: A, B, and C, which

```

// A state is safe iff there exists a safe sequence of grants
// that would allow all threads to eventually receive their
// maximum resource needs
//
bool
ResourceMgr::isSafe()
{
    int j;
    int toBeAvail[] = copy avail[];
    int need[][] = max[][] - alloc[][]; // need[i][j] initialized to max[i][j] - alloc[i][j]
    bool finish[] = {false, false, false, ...}; // finish[j] is true if thread j is guaranteed to finish

    while(true){
        j = any threadID s.t. ( (finish[j]==false) && ( forall i: need[i][j] <= toBeAvail[i]));
        if(no such j exists){
            if (forall j: finish[j] == true){
                return true;
            }
            else{
                return false;
            }
        }
        else{
            // Thread j will eventually finish and return its current
            // allocation to the pool
            finish[j] = true;
            forall i: toBeAvail[i] = toBeAvail[i] + alloc[i][j];
        }
    }
}

// Hypothetically grant request and see if resulting state
// is safe.
//
bool
ResourceMgr::wouldBeSafe(int resourceId, int threadID)
{
    bool ret = false;
    avail[resourceID]--;
    alloc[resourceID][threadID]++;
    if(isSafe()){
        ret = true;
    }
    avail[resourceID]++;
    alloc[resourceID][threadID]--;
    return ret;
}

```

Figure 6.11: Banker's algorithm test on system state (pseudo-code.) Figure 6.10 provides the high-level pseudo-code of the banker's algorithm and and Figure 6.9 defines the state on which these tests work.

may need as many as 4, 5, and 5 pages to complete, respectively.

If they take turns requesting one page each, and the system grants requests in order, the system deadlocks, reaching a state where each process is stuck until some other process releases memory:

Process	Allocation											
A	0	1	1	1	2	2	2	3	3	3	wait	wait
B	0	0	1	1	1	2	2	2	3	3	3	wait
C	0	0	0	1	1	2	2	2	wait	wait	wait	wait
Total	0	1	2	3	4	5	6	7	8	8	8	8

On the other hand, if the system follows the Banker's algorithm, then it can delay some processes and guarantee that all processes eventually complete.

E.g.,

Process	Allocation											
A	0	1	1	1	2	2	2	3	3	3	4	0
B	0	0	1	1	1	2	2	2	wait	wait	wait	3
C	0	0	0	1	1	1	2	2	2	wait	wait	3
Total	0	1	2	3	4	5	6	7	8	4	6	7

By delaying B and C in the ninth through twelfth steps, the algorithm allows A to complete and release its resources. Then, by delaying C in the fifteenth and sixteen steps, the algorithm allows B to complete and release its resources.

Though not terribly complex in absolute terms, the Banker's Algorithm is noticeably more involved than the other approaches discussed above. It is rarely used in its full generality, but understanding the distinction between *safe*, *unsafe*, and *deadlock* states and understanding how manifestations of deadlock depend on request ordering are both important for understanding deadlock.

Additionally, an understanding of the Banker's Algorithm can help one design simple solutions for specific problems. For example, if we apply the Banker's Algorithm to the variation of the Dining Philosopher's problem where we place the chopsticks in the middle of the table, then a philosopher taking a chopstick would wait if it would be the last chopstick and no philosopher would have two chopsticks; otherwise, the philosopher would take the chopstick.

6.2.4 Detecting and breaking deadlock

Rather than preventing deadlocks, some systems allow deadlocks to occur and break them when they arise.

Why allow deadlocks to occur at all? Sometimes, it is difficult or expensive to enforce sufficient structure on the system's data and workloads to prevent deadlock. For example, this approach is often used in databases, which provide a general interface that applications can use to access shared data via multi-step *transactions* that can acquire and release locks covering different subsets of data. Because the database is meant as a general tool, there is seldom any way to prevent users from issuing requests that could cause deadlock. Also, because a database is often shared by multiple users, it can not allow bugs

in one user's queries or unexpected interactions between two users' queries to create deadlocks that permanently make some data inaccessible.

For this approach to work, we need ways to break deadlocks when they occur, ideally with minimal harm to programs, and we need ways to detect deadlocks so that we know when to invoke the recovery mechanisms. We now talk about each of these challenges:

Breaking deadlocks

Breaking a deadlock once it has occurred generally requires forcibly taking resources away from some or all of the deadlocked threads. Because the resources by definition are not revokable, this process generally damages the victim threads in some way, but it hopefully allows the rest of the system to continue to function.

As a simple example, when some operating systems decide that a process is part of a deadlock, they simply kill the process and release the process's resources. Although this sounds drastic, if a deadlocked process cannot make any progress anyhow, killing it does not make it much worse off.

Notice, however, that under the lock-based shared object programming abstractions we have discussed, it is seldom possible to kill deadlocked threads within a process and allow the other threads sharing the process's shared objects to continue to function. If the deadlocked threads hold locks on shared objects, simply killing the threads and marking the locks as free could leave the objects in an inconsistent state.

Transactions. To allow deadlocks to be broken with minimal disruption in systems that use locks, we would like to do two things.

- First, we would like to ensure that revoking locks from one thread does not leave the system's objects in an inconsistent state. To do this, we would like to be able to *undo* a deadlocked thread's actions. Then, to fix a deadlock, we can choose one or more victim threads, stop them, undo their actions, and let other threads proceed.
- Second, once the deadlock is broken and other threads have completed some or all of their work, we would like to be able to *restart* the victim threads. If these threads can now complete, the system operates as if the victim threads never caused a deadlock but, instead, just had the start of their executions delayed.

Transactions, which we discuss in detail in Chapter ??, are widely used in databases and provide these two properties.

A transaction is like a critical section, and it has `beginTransaction` and `endTransaction` statements that are similar to a critical section's `Lock::Acquire()` and `Lock::Release()`. In particular, in that transactions are *isolated* so that during a transaction, the one transaction's actions cannot affect other transactions. Isolation can be ensured with appropriate locking. If all of a transaction's actions including the `endTransaction` complete, then the transaction *commits*, the transaction's locks are released, and the transaction's operations become visible.

One key difference between transactions and critical sections is that transactions can *abort* and *roll back* their actions. If a transaction fails to reach its `endTransaction` statement (e.g., because of a deadlock or because some other exception occurred), the system can reset all of the state modified by the transaction to what it was when the transaction began. One way to support this is to maintain an *undo log* that keeps track of the initial values of all state modified by each transaction.

So, if a transactional system becomes deadlocked, the system can abort one or more of the deadlocked transactions. Aborting these transactions rolls back the system's state to what it would have been if these transactions had never started and releases the aborted transactions' locks and other resources. If aborting the chosen transactions releases sufficient resources, the deadlock is broken and the remaining transactions can proceed. If not, then the system can abort additional transactions.

Since aborting a transaction rolls back the state of the system to what it would have been had the transaction not begun execution, the system can restart the aborted transactions at some later time. For example, a conservative system might minimize the risk of encountering the same deadlock by waiting for all of the current (non-aborted) transactions to complete and then restarting and completing one aborted victim transaction at a time until all of them complete. Alternatively, a more aggressive system could restart multiple victim transactions at the same time, repeating the recovery process if it gets unlucky and deadlocks again.

Definition: **Optimistic concurrency control**

Transactions with optimistic concurrency control. Instead of using transactions as a way to let us break deadlocks, we can also use transactions to avoid them. *Optimistic concurrency control* allows transactions to execute in parallel without locking any data, but it only allows a transaction to commit if none of the objects accessed by the transaction have been modified since the transaction began; otherwise, the transaction must abort. Typically, systems then retry the aborted transaction—since each reexecution runs with a different set of concurrent transactions, it should eventually be able to successfully commit.

One way to implement transactions with optimistic concurrency control is for each transaction to keep track of which versions of which objects it reads and for it to apply its updates to a local copy of each object it modifies. Then,

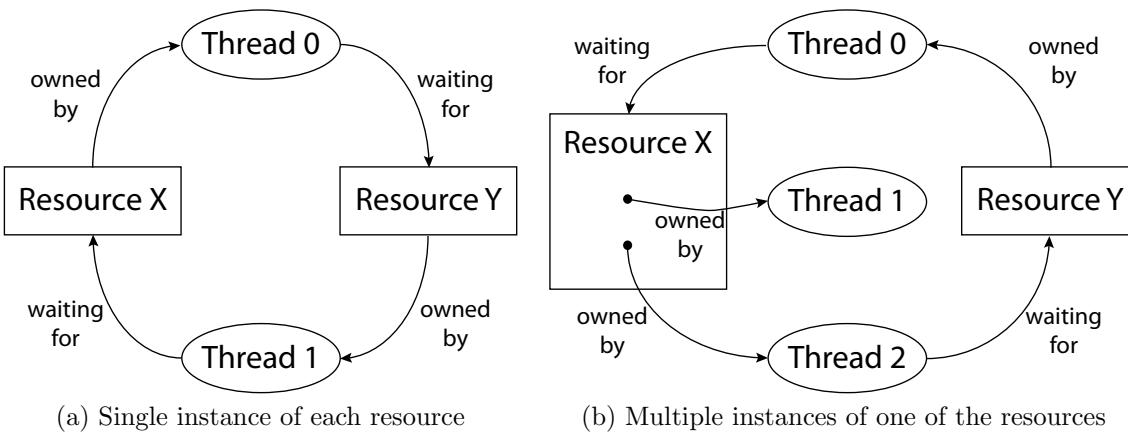


Figure 6.12: In this graph used for deadlock detection, threads and resources are nodes, and directed edges represent the *owned by* and *waiting for* relationships among them.

before a transaction commits, it verifies that none of the objects it accessed were modified by other transaction.

Optimistic concurrency control works well for most systems and workloads, where different transactions usually access different subsets of data. In these cases, the approach not only eliminates deadlock but also maximizes concurrency since threads do not wait for locks. On the other hand, if there are significant numbers of conflicting, concurrent transactions, overheads from rolling back and reexecuting transactions can be high.

Detecting deadlock

There are various ways to detect deadlock.

One simple approach used in some systems is to assume that any thread that fails to make progress is part of a deadlock. This approach risks false positives where a non-deadlocked thread is incorrectly classified as deadlocked, but for some systems an occasional false positive may be an acceptable price to pay for the simplicity of the approach.

If there are several resources and only one thread can hold each resource at a time (e.g., one printer, one keyboard, and one audio speaker or several mutual exclusion locks), then we can detect a deadlock by analyzing a simple graph where each thread and each resource is represented by a node and where there is a directed edge from a resource to a thread if the resource is owned by the thread and a directed edge from a thread to a resource if the thread is waiting for the resource. See, for example, Figure 6.12-a. There is a deadlock if and only if there is a cycle in such a graph.

```

// A state is safe iff there exists a safe sequence of grants
// that would allow all threads to eventually receive their
// maximum resource needs
//
bool ResourceMgr::isDeadlock()
{
    // avail[] holds free resource count;
    // alloc[][] holds current allocation
    // request[][] holds currently-blocked requests
    int j;
    int toBeAvail[] = copy avail[];
    bool finish[] = [false, false, false, ...]; // finish[j] is true if thread j is guaranteed to finish

    while(true){
        j = any threadID s.t. ( (finish[j]==false) && ( forall i: request[i][j] <= toBeAvail[i]) );
        if(no such j exists){
            if (forall j: finish[j] == true){
                return false;
            }
            else{
                return true;
            }
        }
        else{
            // Thread j *may* eventually finish and return its current
            // allocation to the pool
            finish[j] = true;
            forall i: toBeAvail[i] = toBeAvail[i] + alloc[i][j];
        }
    }
}

```

Figure 6.13: Coffman et al.'s algorithm test for deadlock. This algorithm is similar to the `isSafe()` test of the banker's algorithm shown in Figure 6.11. We omit a detailed description of Coffman's high level design and state declarations; these are straightforward variations of the corresponding pseudo-code for the Banker's Algorithm; see Figure 6.10 and 6.9.

If there are multiple instances of some resources, then we represent a resource with k interchangeable instances (e.g., k equivalent printers) as a node with k connection points (e.g., see Figure 6.12-b). Now, a cycle is a necessary but not sufficient condition for deadlock.

Another solution, described by Coffman, Elphick, and Shoshani in 1971 is a variation of Dijkstra's Banker's Algorithm. Now, we assume we no longer know `max[] []`, so we cannot assess whether the current state is safe or whether some future sequence of requests can force us to deadlock. However, we can look at the current set of resources, granted requests, and pending requests and ask whether it is possible for the current set of requests to eventually be satisfied assuming no more requests come and all threads eventually complete. If so, there is no deadlock (yet, though we may be in an unsafe state); otherwise, there is a deadlock.

Figure 6.13 shows the pseudocode of the main `testIfDeadlocked()` method,

a variation of the `isSafe()` method shown in Figure 6.11 for the Banker’s Algorithm. We leave the state variables and high level interface (Figures 6.9 and 6.10) as an exercise for the reader.

One might hope that we could avoid deadlock by asking “Will satisfying the current request put us in a deadlocked state?” and then blocking requests that do, but as this algorithm highlights, deadlock is determined not just by what requests we grant but also by what requests are waiting. The request that puts us into a deadlocked state (“circular wait”) will be a request that waits, not a request that is granted.

Taking a step back, it is the lack of knowledge about possible future requests prevents us from using this algorithm to avoid deadlock. As Figure 6.8 illustrates, we can be in an unsafe state long before we reach a deadlock state, and once we reach an unsafe state, there are request sequences that will force us to deadlock.

For example, recall the ACB/BCA example on page 265. Even though we are not yet deadlocked after thread 1 acquires A and thread 2 acquires B, once these two actions occur, deadlock is inevitable given the requests that will arrive in the future.

Exercises

1. Figure 6.2 shows an execution that executes some requests in parallel, and it shows an equivalent sequential execution—request 1 then request 2 then request 3—. There are two other sequential executions that are also equivalent to the parallel execution shown in the figure. What are these other equivalent sequential executions?
2. Generalize the rules for two phase locking to include both mutual exclusion locks and readers-writers locks. What can be done in the expanding phase? What can be done in the contracting phase?
3. Consider the variation of the Dining Philosophers problem shown in Figure 6.7 where all unused chopsticks are placed in the center of the table and any philosopher can eat with any two chopsticks.
One way to prevent deadlock in this system is to provide sufficient resources. For a system with n philosophers, what is the minimum number of chopsticks that ensures deadlock freedom? Why?
4. If the queues between stages are finite, Is it possible for a staged architecture to deadlock even if each individual stage is internally deadlock free? If so, give an example. If not, prove it.

5. Suppose you build a system using a staged architecture with some fixed number of threads operating on stage. Assuming each stage is individually deadlock free, describe two ways to guarantee that your system as a whole can not deadlock. Each of the ways should eliminate a different one of the 4 necessary conditions for deadlock.

6.3 Alternative approaches to synchronization

Chapter 5 described a core abstraction for synchronization—shared objects with one lock per object. This abstraction is the right building block for multi-threaded programs the vast majority of the time. Occasionally, as start of the current chapter indicated, you need to resort to fine grained locking, a variation of this basic approach that divides an object’s state among different locks.

Even more rarely programmers resort to alternatives that avoid locks such as *read-copy-update (RCU)* synchronization and *lock-free* and *wait-free data structures*. We emphasize that the cases when these approaches are warranted are rare and that these advanced techniques should only be considered by experienced programmers who have mastered the basic lock-based approaches. Many readers will probably never need to use these techniques. If you do find yourself tempted to do so, take extra care. Be sure to measure the performance of your system to make sure that these techniques yield significant gains, and seek out extra peer review from trusted colleagues to help make sure that the code works as intended.

We caution that programmers are often tempted to assume that acquiring a lock is an expensive operation and to therefore try to reduce locking throughout their programs. The most likely result from this premature optimization mindset is a program that is buggy, hard to maintain, no faster than a clean implementation, and—ironically—harder to tune than a cleanly architected program. On most platforms, acquiring or releasing a lock is a highly tuned primitive—acquiring an uncontended lock is often nearly free (and if there is contention then you probably need that lock!)

That said, although you may not often (or ever) write code that uses these advanced techniques, it is important to understand them because they do get used in critical parts of important systems such as the Linux kernel and some Java Virtual Machine libraries and because in some cases they can provide significant performance gains.

6.3.1 Read-Copy-Update (RCU)

The goal of read-copy-update (RCU) is to provide high performance synchronization for data structures that are frequently read and occasionally updated.

In particular, RCU optimizes the read path to have extremely low synchronization costs, even if there are many concurrent readers. However, writes can be delayed for a long time—tens of milliseconds in some implementations.

Why RCU? We might hope that readers-writers locks would be an adequate solution for read-dominated workloads. Recall that readers-writers locks allow multiple concurrent active readers, but there is an active writer no other writer or reader can be active.

The problem with the readers-writers locks approach is its cache behavior for concurrent reads with short critical sections. Recall that before reading, a reader acquires a readers-writers lock in read mode. To do this, it must read and then update some state in the readers-writers synchronization object. Unfortunately, this access pattern causes a large number of cache misses when there are a large number of concurrent readers.

In particular, on a multiprocessor, when one processor updates a hardware cache line, it invalidates that cache line in all other caches. Then, when another processor wants to read and update the data, it first suffers a cache miss and must fetch the data from the first processor’s cache or from main memory; then it must invalidate the cache line in other caches; finally, it can update the data in its cache. On a modern processor, fetching and invalidating a cache line can take hundreds of cycles. If there are a large number of processors trying to read a data structure protected by a readers-writers lock, the average processor may wait thousands of cycles to acquire the lock in read mode, even if there are no writers.

So, for critical sections less than a few thousand cycles and for programs where there may be many threads simultaneously trying to read an object, the standard readers-writers lock can impose significant overheads.

The RCU approach. How can we let concurrent reads access a data structure that can also be written without having to suffer the cache effects of updating the state of a synchronization variable on each read?

To meet this challenge, RCU weakens its semantics in two ways compared to readers-writers locks.

1. **Relax R/W semantics.** RCU allows up to one read/write critical section to be concurrent with any number of read-only critical sections. A read-only critical section that overlaps a read/write critical section may see the old or new version of the data structure.

Note that an object that uses RCU for synchronization must maintain multiple versions of its state, and it must guarantee that an old version is not freed until all readers have finished accessing it. The time from when an update occurs until the old version has been freed is called the *grace*

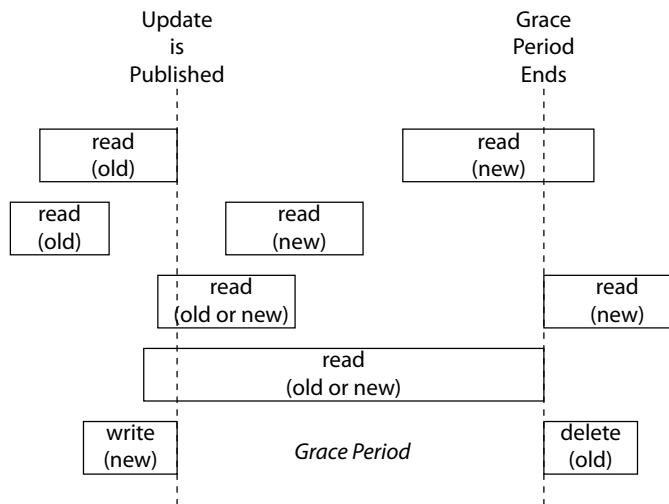


Figure 6.14: Timeline for an update concurrent with several reads for a data structure accessed with read-copy-update (RCU) synchronization.

period, and RCU must provide a way to determine when a grace period Definition: grace period ends.

2. **Restrict update rules.** To make a data structure update appear atomic to readers, an RCU update must be *published* to the data structure with a single, atomic memory write (e.g., by updating a single pointer.)

Figure 6.14 shows the timeline for a write critical section that is concurrent with several read critical sections under RCU. If a function that reads the data structure completes before a write is published, it will (of course) see the old version of the data structure, and if a read critical section begins after a write is published it will see the new version. But, if a read critical section begins before and ends after a write is published, it may see the old version or the new one; if it reads the updated pointer more than once, it may even first see the old one and then see the new one. Which version it sees depends on which version of the single atomically-updated memory location it observes. Furthermore, the system guarantees that the old version is not deleted until the grace period expires, which means that deletion of the old versions must be delayed until all reads that might be observing the old version have completed.

Definition: RCU API and use. *RCU* is a synchronization abstraction that allows concurrent access to a data structure by multiple readers and a single writer at a time. Figure 6.15 shows a typical API.

A reader calls `RCU::ReadLock()` and `RCU::ReadUnlock()` before and after accessing the shared data structure. A writer calls `RCU::WriteLock()` to ex-

Reader API	RCU::ReadLock()
	RCU::ReadUnlock()
Writer API	RCU::WriteLock()
	RCU::Publish()
	RCU::WriteUnlock()
	RCU::Synchronize()

Figure 6.15: API for read-copy-update (RCU) synchronization.

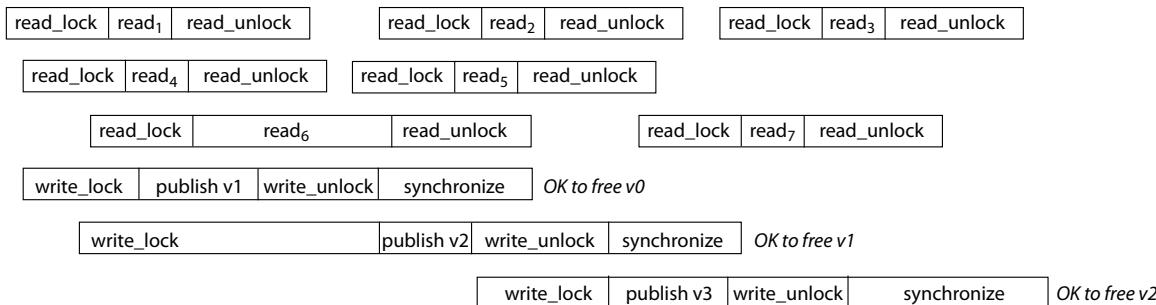


Figure 6.16: RCU allows one write at a time, and it allows reads to overlap each other and writes. The initial version is v0, and overlapping writes update the version to v1, v2, and then v3.

clude other writers, RCU::Publish() to issue the write that atomically updates the data structure so that reads can see the updates, RCU::WriteUnlock() to allow other writers to proceed, and RCU::Synchronize() to wait for the grace period to expire so that the old version of the object can be freed. Notice that as Figure 6.16 illustrates, writes are serialized—only one write can proceed at a time. But also notice that writes can be concurrent with reads and that one write’s update can be concurrent with another write’s grace period: there may be any number of versions of the object until multiple overlapping grace periods expire.

Example: Overlapping operations.

Question: For each read in Figure 6.16, which version(s) of the shared state can the read observe?

Answer: If a read overlaps a publish, it can return the published value or the previous value.

```

typedef struct ElementS{
    int key;
    int val1;
    int val2;
    struct ElementS *next;
} Element;

class RCULL{
private:
    RCULock rcuLock;
    Element *head;
public:
    int search(int key, int *ret1, int *ret2);
    void insert(Element *item);
    void remove(int key);
};

```

Figure 6.17: Declaration of data structures and API for a linked list that uses RCU for synchronization.

<i>read</i> ₁	v0 or v1	overlaps publish v1
<i>read</i> ₂	v2	after publish v2, before publish v3
<i>read</i> ₃	v3	after publish v3
<i>read</i> ₄	v0 or v1	overlaps publish v1
<i>read</i> ₅	v1 or v2	overlaps publish v2
<i>read</i> ₆	v0, v1, or v2	overlaps publish v1 and v2
<i>read</i> ₇	v3	after publish v2

Example: RCU linked list. Suppose we have linked list of records, such as the one defined in Figure 6.17. Here, the list comprises an RCU lock `rcuLock` and a pointer to the head of the list `head`. Each record has three data fields, `key`, `val1`, and `val2`, and a `next` pointer to the next record on the list.

Figure 6.19 shows how we implement a `search()` method that scans down the list until an element with a matching key is found. If such an element is found, the method returns the two value fields by setting the two result pointers and providing a nonzero return value. Otherwise, the method returns 0 to indicate that no matching record was found. Since this method does not modify any of the list’s state, it acquires and releases the RCU lock in read mode.

Figure 6.19 shows two methods that update the list. Each of them is arranged so that a single pointer update is all that is needed to publish the list update to the readers. In particular, notice that it is important that `insert()` initialize the data structure *before* updating the `head` pointer to make the new element visible to readers.

Implementing RCU. In implementing RCU, the central goal is to minimize the cost of read critical sections—the cost should be low and should be constant

```

int
RCULL::search(int key, int *ret1, int *ret2){
    rcuLock.ReadLock();
    int ret = 0;
    Element *current = head;
    while(current != NULL){
        if(current->key == key){
            *ret1 = current->val1;
            *ret2 = current->val2;
            ret = 1;
            current = NULL; // break out of loop
        }
    }
    rcuLock.ReadUnlock();
    return ret;
}

```

Figure 6.18: Implementation of a read-only method for a linked list that uses RCU for synchronization.

```

void
RCULL::remove(int key){
    // One write at a time
    rcuLock.WriteLock();

void
RCULL::insert(int key, int val1, int val2){
    // One write at a time
    rcuLock.WriteLock();

    // Initialize item
    Element *item;
    item = (Element*)malloc(sizeof(Element));
    item->key = key;
    item->val1 = val1;
    item->val2 = val2;
    item->next = head;

    // Atomically update list
    rcuLock.Publish(&head, item);

    // Allow other writes to proceed
    rcuLock.WriteUnlock();

    // On return, no reader has old version
    rcuLock.Synchronize();
}

int found = 0;
Element *prev=NULL;
Element *current = head;
while(!found && current != NULL){
    if(current->key == key){
        found = 1;
        // Publish update to readers
        if(prev != NULL){
            rcuLock.Publish(&(prev->next),
                           current->next);
        }
        else{
            rcuLock.Publish(&head,
                           current->next);
        }
    }
}
// Allow other writes to proceed
rcuLock.WriteUnlock();
if(found){
    // Wait until no reader has old version
    rcuLock.Synchronize();
    free(current);
}
}

```

Figure 6.19: Implementation of two updating methods for a linked list that uses RCU for synchronization.

```

class RCULock{
    private:
        // Global state
        Spinlock globalSpin;
        long globalCounter;
        DEFINE_PER_PROCESSOR(static long, quiescentCount); // One per processor

        // Per-lock state
        Spinlock writerSpin;

        // Public API omitted
    ...
}

```

Figure 6.20: Global and per-lock data structures for a simple quiescence-based RCU implementation. Credit: This pseudocode is based on an implementation by Paul McKenney in “Is Parallel Programming Hard, And, If So, What Can be Done About It?”

regardless of the number of concurrent readers. Conversely, we are willing to allow writes to have a high *latency* (i.e., we are willing to allow long grace periods, where it might be tens of milliseconds from when an update is published until the system can guarantee that no readers can access the old version), though we would like the *overhead* of writes to stay modest (i.e., the CPU consumption of the write operations should be small, allowing a large numbers of writers to complete their updates quickly, even if old versions may continue to be used by readers for some time, and allowing other threads to run while a thread waits for a grace period to expire.)

A common technique for achieving these goals is to change the scheduler, integrating the RCU implementation with that of the scheduler. In contrast with the synchronization primitives described in the previous chapter, which make no assumptions about the scheduler, this integration allows us to rely on strong assumptions about when reads can occur, which relieves us of the (expensive) responsibility of tracking exactly how many readers are active at any given time. In particular, our implementation will require two things from the scheduler: (1) RCU read critical sections complete without being interrupted and (2) whenever a thread on a processor is interrupted, the scheduler updates some per-processor RCU state. Then, once a write completes, `RCULock::Synchronize()` simply waits for all processors to be interrupted at least once. At that point, the old version of the object is known to be *quiescent*—no thread has access to that old version (other than the writer that changed it.)

Definition: **quiescent**

Figure 6.20 and Figure 6.21 show a simple implementation of RCU based on quiescent states.

The first thing to notice is that `ReadLock()` and `ReadUnlock()` are extremely cheap operations—they update no state and merely ensure that the read will not be interrupted; in a non-preemptable kernel, these functions need to do

```

void RCULOCK::WriteLock(){
    writerSpin.Acquire(); // Includes MEMORY_BARRIER
}

void RCULOCK::ReadLock(){
    // No-op in nonpreemptable kernel
    DISABLE_INTERRUPTS();
}

void RCULOCK::ReadUnlock(){
    // No-op in nonpreemptable kernel
    ENABLE_INTERRUPTS();
}

// Called by scheduler
void RCULOCK::QuiescentState(){
    MEMORY_BARRIER();
    PER_PROC_VAR(quiescentCount) =
        globalCounter;
    MEMORY_BARRIER();
}

void RCULOCK::WriteUnlock(){
    writerSpin.Release(); // Includes MEMORY_BARRIER
}

void RCULOCK::Publish(void **pp1, void *p2){
    *pp1 = p2;
    MEMORY_BARRIER();
}

void
RCULOCK::Synchronize(){
    int p, c;
    globalSpin.Acquire(); // Includes MEMORY_BARRIER
    c = ++globalCounter;
    globalSpin.Release(); // Includes MEMORY_BARRIER
    FOREACH_PROCESSOR(p){
        while(PER_PROC_VAR(quiescentCount, p) - c < 0){
            sleep(10); // Release CPU for 10ms
        }
    }
}

```

Figure 6.21: A simple quiescence-based RCU implementation. Note: We assume that `SpinLock::Acquire()` and `SpinLock::Release()` each include a `MEMORY_BARRIER()`, though such memory barriers are not shown in our earlier `SpinLock` implementation. Credit: This pseudocode is based on an implementation by Paul McKenney in “Is Parallel Programming Hard, And, If So, What Can be Done About It?”

nothing at all.

`WriteLock()` and `WriteUnlock()` simply acquire and release a spinlock. This ensures that at most one write per `RCULOCK` can proceed at a time.

`Publish()` is also simple. It replaces the specified pointer with a new one and then executes a memory barrier so that threads on all processors will observe the update.

`Synchronize()` and `QuiescentState()` work together to ensure that when `Synchronize()` returns, all threads are guaranteed to be done with the old version of the object. `Synchronize()` increments a global counter `globalCounter` and then waits until all processors’ `quiescentCounts` match `c`, the new value of that counter. `QuiescentState()`, which is called by the scheduler whenever it interrupts a running thread, updates the interrupted processor’s `quiescentCount` by setting it to match the current `globalCounter`. Thus, once a call to `Synchronize()` determines that all processors’ `quiescentCounts` are at least as large as `c`, it knows that all processors have stopped running any read critical sections that could have been concurrent with the update, and that no remaining read critical sections can observe the old version.

6.3.2 Lock-free and wait-free data structures

RCU allows reads to proceed without acquiring a lock or updating shared synchronization state, but it still requires updates to acquire locks. If the thread that holds the lock is interrupted, has a bug that causes it to stop making progress, or becomes deadlocked, other threads can be delayed for a long—perhaps unlimited—period of time.

It is possible to build data structures that completely avoid locking on both reads and writes. *Lock free* data structures ensure eventual progress: if all threads are allowed to run for some finite number of steps, then at least one of the threads will complete its operation. *Wait free* data structures ensure both eventual progress and fairness: if any thread is allowed to run for some finite number of steps, that thread will complete its operation regardless of the execution speed of the other threads.

Historically the design of efficient lock-free or wait-free data structures has been complex and application-specific. Nonetheless, lock free or wait free algorithms of varying levels of efficiency exist for wide range of data structures including FIFO queues, double-ended queues, LIFO stacks, sets, and hash tables.

Designing efficient wait-free data structures continue to be the domain of experts, and a discussion of the techniques they use is outside the scope of this book. However, some reasonably-efficient and general approaches to implementing lock-free data structures are known and in sufficiently wide use to warrant further comment here.

Lock-free data structures. Lock free data structures can be implemented by having each operation detect concurrent, conflicting operations. When such updates are detected, the operation waits a finite amount of time for them to finish. If the conflicting operations do not finish, the operation either aborts the conflicting ones (rolling the object's state back to what it was before that conflicting operations began) or it assists them (finishing the updates needed to complete that operation).

Example: Transactions and software transactional memory (STM)
A transactional database with optimistic concurrency control, is an example of a very flexible, lock-free data structure capable of running programs supplied by different users accessing arbitrary data in arbitrary ways without risking deadlock. Recall that optimistic concurrency control allows transactions to proceed without locking the data they access and aborts the transaction if at commit-time any of the accessed data have changed.

To see that such databases are lock free, consider two conflicting transactions executing at the same time. The first one to commit will succeed, and the second one must abort and retry.

On the other hand, the approach is not wait-free since it cannot bound the number of retries needed for a transaction to successfully commit—in the worst case, it is possible for a given transaction to starve forever if it encounters a stream of conflicting transactions that always manage to beat it in the race to commit.

Although much work on transactions has been done in the context of databases that store data on disk, *software transactional memory (STM)* is a promising approach to allow transactions on in-memory data structures. Unfortunately, the cost of an STM transaction is often significantly higher than that of a traditional critical section because of the need to maintain the state needed to check dependencies and the state needed either to update the object if there is no conflict or to roll back its state if a conflict is detected. On the other hand, in situations where STM can be used, it provides a way to compose different modules without having to worry about deadlock.

Definition: **software transactional memory (STM)**

Exercises

6. In `RCUlist::remove()`, suppose we attempt to maximize concurrency by replacing the `WriteLock()` and `WriteUnlock()` calls with `ReadLock()` and `ReadUnlock()` calls and insert new `WriteLock()` and `writeUnlock()` calls at beginning and end of the code that is executed only if the `if` conditional test succeeds. The basic idea is to hold a read lock while searching for the target item and to grab the write lock once it is found. Will this work?

6.4 Conclusion

The quotes introducing this chapter are intended to emphasize that advanced synchronization techniques should be approached with caution. Your first goal should be to construct a program that works, even if doing so means putting “one big lock” around everything in a data structure or even in an entire program.

Resist the temptation to do complicated fine grained locking (let alone RCU) unless you **know** that doing so is necessary. How do you know? Don’t guess. Measure your system’s performance. (Measuring the “before” and “after” performance of a program and its subsystems not only helps you make good decisions about the program on which you are working, but it also will help you develop good intuition for the programs you face in the future.)

Spend a lot of time early in the design process developing a clean structure for your program. Given that multi-object synchronization and deadlock are

not modular, it is vital to have an overall structure that lets you reason about how the pieces will interact. Often, it is helpful to strive for a strict layering or hierarchy of modules. It is easier to make such programs deadlock free, and it is easier to test them as well.

Performance is important, but it is usually easier to start with a clean, simple, and correct design, measure it to identify its bottlenecks, and then optimize the bottlenecks than to start with a complex design and try to tune its performance (let alone fix its bugs.)

Problems

Exercises

1. Figure 6.2 shows an execution that executes some requests in parallel, and it shows an equivalent sequential execution—request 1 then request 2 then request 3. There are two other sequential executions that are also equivalent to the parallel execution shown in the figure. What are these other equivalent sequential executions?
2. Generalize the rules for two phase locking to include both mutual exclusion locks and readers-writers locks. What can be done in the expanding phase? What can be done in the contracting phase?

3. Consider the variation of the Dining Philosophers problem shown in Figure 6.7 where all unused chopsticks are placed in the center of the table and any philosopher can eat with any two chopsticks.

One way to prevent deadlock in this system is to provide sufficient resources. For a system with n philosophers, what is the minimum number of chopsticks that ensures deadlock freedom? Why?

4. If the queues between stages are finite, Is it possible for a staged architecture to deadlock even if each individual stage is internally deadlock free? If so, give an example. If not, prove it.
5. Suppose you build a system using a staged architecture with some fixed number of threads operating on stage. Assuming each stage is individually deadlock free, describe two ways to guarantee that your system as a whole can not deadlock. Each of the ways should eliminate a different one of the 4 necessary conditions for deadlock.

6. In `RCUlist::remove()`, suppose we attempt to maximize concurrency by replacing the `WriteLock()` and `WriteUnlock()` calls with `ReadLock()` and `ReadUnlock()` calls and insert new `WriteLock()` and `writeUnlock()` calls at beginning and end of the code that is executed only if the `if` conditional test succeeds. The basic idea is to hold a read lock while searching for the target item and to grab the write lock once it is found. Will this work?

Chapter 7

Scheduling

Time is money – Ben Franklin

The best performance improvement is the transition from the non-working state to the working state. That's infinite speedup. – John Ousterhout

When there are multiple things to do, how do you choose which one to do first? In the last few chapters, we have described how to create threads, switch between them, and synchronize their access to shared data. At any point in time, some threads are running on the processor. Others are waiting their turn for the processor. Still other threads are blocked waiting for I/O to complete, a condition variable to be signalled, or for a lock to be released. When there is more than one runnable thread, the *processor scheduling policy* determines which thread to run next.

Definition: **processor scheduling policy**

You might think the answer to this question is easy: just do the work in the order in which it arrives. After all, that seems to be the only fair thing to do. Because it is obviously fair, almost all government services work this way. When you go to your local Department of Motor Vehicles (DMV) to get a driver's license, you take a number and wait your turn. Although fair, the DMV often feels slow. There's a reason why: as we'll see later in this chapter, doing things in order of arrival is sometimes the worst thing you can do in terms of improving user-perceived response time. Advertising that your operating system uses the same scheduling algorithm as the DMV is probably not going to increase your sales!

You might think that the answer to this question is unimportant. With the million-fold improvement in processor performance over the past thirty years, it might seem that we are a million times less likely to have anything waiting for its turn on the processor. We disagree! Server operating systems in particular are often overloaded. Parallel applications can create more work than processors, and if care isn't taken in the design of the scheduling policy, performance can badly degrade. There are subtle relationships between scheduling policy and

energy management on battery-powered devices such as smartphones and laptops. These questions apply whether the source of contention is the processor, memory, disk, or network, and so we will revisit them throughout the rest of this book.

Scheduling policy is not a panacea. Without enough capacity, performance may be poor regardless of which thread we run. In this chapter, we will also discuss how to predict overload conditions and how to adapt to them.

Fortunately, you probably have quite a bit of intuition as to impact of different scheduling policies and capacity on issues like response time, fairness, and throughput. Anyone who waits in line probably wonders how we could get the line to go faster. That's true whether we're waiting in line at the supermarket, a bank, the DMV, or at a popular restaurant. Remarkably, in each of these settings, there is a different approach to how they deal with waiting. We'll try to answer why.

There is no one right answer; rather, any scheduling policy poses a complex set of tradeoffs between various desirable properties. The goal of this chapter is not to enumerate all of the interesting possibilities, explore the full design space, or even to identify specific useful policies. Instead, we describe some of the trade-offs and try to illustrate how a designer can approach the problem of selecting a scheduling policy.

Consider what happens if you are running the web site for a company trying to become the next Facebook. Based on history, you'll be able to guess how much server capacity you need to be able to keep up with demand and still have reasonable response time. What happens if your site appears on Slashdot, and suddenly you have twice as many users as you had an hour ago? If you aren't careful, everyone will think your site is terribly slow, and permanently go elsewhere. Google, Amazon, and Yahoo have each estimated that they lose approximately 5-10% of their customers if their response time increases by as little as 100 milliseconds.

- Would quickly implementing different scheduling policy help, or hurt?
- How much worse will your performance be if the number of users doubles again?
- Should you turn away some users so that others will get acceptable performance?
- Does it matter which users you turn away?
- If you run out to the store and buy a server, how much better will performance get?
- Do the answers change if you are under a denial-of-service attack by a competitor?

Performance Terminology

In Chapter 1 we defined some performance-related terms we will use throughout this chapter and the rest of the book; we summarize those terms here.

- **Task.** A user request. A *task*, sometimes called a *job*, can be any size, from skat-ing the mouse across the screen to computing the shape of a newly discovered protein. When discussing scheduling, we use the term task, rather than thread or process, because a single thread may be responsible for multiple user requests, e.g., in a word processor, each character typed is an individual user request to add that character to the file and display the result on the screen.
 - **Response time (or delay).** The user-perceived time to do some task.
 - **Predictability.** Low variance in response times for repeated requests.
 - **Throughput.** The rate at which tasks are completed.
 - **Scheduling overhead.** The time to switch from one task to another.
 - **Fairness.** Equality in the number and timeliness of resources given to each task.
 - **Starvation.** The lack of progress for one task, due to resources given to a higher priority task.
-
-

In this chapter, we will try to give you the conceptual and analytic tools to help you answer these questions.

The key topics in this chapter are:

- **Uniprocessor scheduling.** How do uniprocessor scheduling policies affect fairness, response time and throughput?
- **Multiprocessor scheduling.** How do scheduling policies change when we have multiple processor cores per computer?
- **Energy-efficient and deadline scheduling.** Many new computer systems allow energy to be saved by running computations more slowly. How do we make this tradeoff, while still minimizing the impact on user perceived response time? More generally, how do we make sure our jobs finish in time?
- **Queueing theory.** In a server environment, how are response time and throughput affected by the rate at which requests arrive for processing and by the scheduling policy?
- **Overload control.** How do we keep response time reasonable when a system becomes overloaded?

7.1 Uniprocessor scheduling

We start by considering one processor, generalizing to multiprocessor scheduling policies in the next section. We begin with three simple policies — first-in-first-out, shortest-job-first, and round robin — as a way of illustrating scheduling concepts. Each approach has its own strengths and weaknesses, and most resource allocation systems (whether for processors, memory, network or disk) combine aspects of all three. At the end of the discussion, we will show how the different approaches are synthesized into a more practical and complete processor scheduler.

Definition: **workload**

Before proceeding, we need to define a few terms. A *workload* is a set of tasks for some system to perform, along with when each task arrives and how long each task takes to complete. In other words, the workload defines the input to a scheduling algorithm. Given a workload, a processor scheduler decides when each task is to be assigned the processor.

Definition: **compute-bound**

We are interested in scheduling algorithms that work well across a wide variety of environments, because workloads will vary quite a bit from system to system and user to user. Some tasks are *compute-bound* and only use the processor. Others, such as a compiler or a web browser, mix I/O and computation.

Definition: **I/O bound**

Still others, such as a BitTorrent download, are *I/O bound*, spending most of their time waiting for I/O and only brief periods computing. In the discussion, we start with very simple compute-bound workloads and then generalize to include mixtures of different types of tasks as we proceed.

Definition: **work-conserving**

Some of the policies we outline are the best possible policy on a particular metric and workload, and some are the worst possible policy. When discussing optimality and pessimality, we are only comparing to policies that are *work-conserving*. A scheduler is work-conserving if it never leaves the processor idle if there is work to do. Obviously a trivially poor policy has the processor sit idle for long periods when there are tasks in the ready queue.

Definition: **preempt**

Our discussion also assumes the scheduler has the ability to *preempt* the processor and give it to some other task. Preemption can happen either because of a timer interrupt, or because some task arrives on the ready list with a higher priority than the current task, at least according to some scheduling policy. We explained how to switch the processor between tasks in Chapters 2 and 4. While much of the discussion is also relevant to non-preemptive schedulers, there are few such systems left, so we leave that issue aside for simplicity.

7.1.1 First In First Out (FIFO)

Perhaps the simplest scheduling algorithm possible is first-in-first-out (FIFO): do each task in the order in which it arrives. (FIFO is sometimes also called first-come-first-served, or FCFS.) When we start working on a task, we keep running it until it is done. FIFO minimizes overhead, switching between tasks

Figure 7.1: Completion times with FIFO and SJF scheduling when short tasks arrive just after a long task.

only when each one completes. Because it minimizes overhead, if we have a fixed number of tasks, and those tasks only need the processor, FIFO will have the best throughput: it will complete the most tasks the most quickly. And as we mentioned, FIFO appears to be the definition of fairness — every task patiently waits its turn.

Unfortunately, FIFO has a weakness. If a task with very little work to do happens to land in line behind a task that takes a very long time, then the system will seem very inefficient. Figure 7.1 illustrates a particularly bad workload for FIFO. If the first task in the queue takes one second, and the next five arrive an instant later, but each only needs a millisecond of the processor, then they will all need to wait until the first one finishes. The average response time will be over a second, but the optimal average response time is much less than that. In fact, if we ignore switching overhead, there are some workloads where FIFO is literally the worst possible policy for average response time.

7.1.2 Shortest Job First (SJF)

If FIFO can be a poor choice for average response time, is there an optimal policy for minimizing average response time? The answer is yes: schedule the shortest job first (SJF).

Suppose we could know how much time each task needed at the processor.

FIFO and memcached

Although you may think that FIFO is too simple to be useful, there are some important cases where it is exactly the right choice for the workload. One such example is memcached. Many web services, such as Facebook, store their user data in a database. The database provides flexible and consistent lookups, such as, which friends need to be notified of a particular update to a user's Facebook wall. In order to improve performance, Facebook and other systems put a cache called memcached in front of the database, so that if a user posts two items to her Facebook wall, the system only needs to do the database lookup the friend list once. The system first checks whether the information is cached, and if so uses that copy.

Because almost all requests are for small amounts of data, memcached is designed to reply to requests in FIFO order. This minimizes overhead, as there is no need to time slice between requests. Because tasks are equal in size, this minimizes both average response time and the variance in response time, and it even maximizes throughput. Win-win!

(In general, we won't know, so this isn't meant as a practical policy! Rather, we use it as a thought experiment; later on, we'll see how to approximate SJF in practice.) If we always schedule the task that has the least remaining work to do, that will minimize average response time. (For this reason, some call SJF shortest-remaining-time-first or SRTF.)

To see that SJF is optimal, consider a hypothetical alternative policy that is not SJF, but that we think might be optimal. At some point this alternative will run a task that is longer than something else in the queue; after all, it is not SJF! If we now switch the order of tasks, keeping everything the same, but doing the shorter task first, we will reduce the average response time. Thus, SJF must be optimal.

Figure 7.1 illustrates SJF on the same example we used for FIFO. If a long task is the first to arrive, it will be scheduled (if we are work-conserving). When a short task arrives a bit later, the scheduler will preempt the current task, and start the shorter one. The remaining short tasks will be processed in order of arrival, followed by finishing the long task.

What counts as "shortest" is the remaining time left on the task, not its original length. If we are one nanosecond away from finishing an hour long task, we will minimize average response time by staying with that task, rather than preempting it for a minute long task that just arrived on the ready queue. Of course, if they both arrive at about the same time, doing the minute long task first will dramatically improve average response time.

Does SJF have any other downsides (other than being impossible to implement because it requires knowledge of the future)? It turns out that SJF is

Starvation and Sample Bias

Systems that might suffer from starvation require extra care when being measured. Suppose you want to compare FIFO and SJF experimentally. You set up two computers, one running each scheduler, and send them the same sequence of tasks. After some period you stop and report the average response time of completed tasks. If some tasks are starved, however, the set of completed tasks will be different for the two policies. Worse, we will have excluded the longest tasks from the results for SJF, skewing the average response time even further. Put another way, if you want to manipulate statistics to prove a point, this is a good trick to use! We leave the solution to this issue to the problem set at end of the chapter.

pessimal for variance in response time. By doing the shortest tasks as quickly as possible, SJF necessarily does longer tasks as slowly as possible (among policies that are work conserving). In other words, there is a fundamental tradeoff between improving average response time and the variance in average response time.

Worse, SJF can suffer from starvation and frequent context switches. If enough short tasks arrive, long tasks may never complete. Whenever a task is put on the ready queue that is shorter than the remaining time left on the currently scheduled task, the scheduler will preempt it. If this keeps happening indefinitely, the long task will never finish.

Suppose a supermarket manager reads a portion of this textbook and decides to implement shortest job first to reduce average waiting times. The manager tells herself: who cares about variance! A benefit is that there would no longer be any need for express lanes — if someone has only a few items, they are immediately whisked to the front of the line, interrupting anyone shopping for eighteen kids. Of course, the wait times of the customers with full baskets skyrocket, so they might decide instead to go to the supermarket down the street, hurting profit margins.

Customers could also try to game the system: if you have a lot of items to purchase, simply go through the line with one item at a time — you will always be whisked to the front, at least until everyone else figures out the same dodge.

Exercises

1. For shortest job first, if the scheduler assigns a task to the processor, and no other task becomes schedulable in the meantime, will the scheduler ever preempt the current task? Why or why not?
2. Devise a workload where FIFO is pessimal — it does the worst possible

Figure 7.2: Completion times with Round Robin scheduling when short tasks arrive just after a long task, with a time quantum of 1ms and 100ms.

choices — for average response time.

3. Suppose you do your homework assignments in SJF-order. After all, you feel like you are making a lot of progress! What might go wrong?

7.1.3 Round robin

Definition: **time quantum**

A policy that addresses starvation is to schedule tasks in a round robin fashion. With Round Robin, tasks take turns running on the processor for a limited period of time. The scheduler assigns the processor to the first task in the ready queue, setting a timer interrupt for some delay, called the *time quantum*. At the end of the quantum, if the task hasn't completed, the task is preempted and the processor is given to the next task in the ready queue. The preempted task is put back on the ready queue where it can wait its next turn. With Round Robin, there is no possibility that a task will starve — it will eventually reach the front of the queue and get its time quantum.

Of course, we need to pick the time quantum carefully. One consideration is overhead: if we have too short a time quantum, the processor will spend all of its time switching and getting very little useful work done. But if we pick too long a time quantum, tasks will have to wait a long time until they get a

What is the overhead of a Round Robin time slice?

One might think that the cost of switching tasks after a time slice is modest: the cost of interrupting the processor, saving its registers, dispatching the timer interrupt handler, and restoring the registers of the new task. On a modern processor, all these steps can be completed in a few tens of microseconds.

However, we must also include the impact of Round Robin on the efficiency of the processor cache. Each new task will need to fetch its data from memory into cache, evicting some of the data that had been stored by the previous task. Exactly how long this takes will depend on the memory hierarchy, the reference pattern of the new task, and whether any of its state is still in the cache after its previous time slice. Modern processors often have multiple levels of cache to improve performance. Reloading just the first level on-chip cache from scratch can take several milliseconds; reloading the second and third level caches takes even longer. Thus it is typical for operating systems to set their time slice interval to be somewhere between 10 and 100 milliseconds, depending on the tradeoff between better responsiveness and reduced overhead.

turn. Figure 7.2 shows the behavior of Round Robin, on the same workload as in Figure 7.1, for two different values for the time quantum.

A good analogy for Round Robin is a particularly hyperkinetic student, studying for multiple finals simultaneously. You won't get much done if you read a paragraph from one textbook, then switch to reading a paragraph from the next textbook, and then switch to yet a third textbook. But if you never switch, you may never get around to studying for some of your courses.

One way of viewing Round Robin is as a compromise between FIFO and SJF. At one extreme, if the time quantum is infinite (or at least, longer than the longest task), Round Robin behaves exactly the same as FIFO. Each task runs to completion and then yields the processor to the next in line. At the other extreme, suppose it was possible to switch between tasks with zero overhead, so we could choose a time quantum of a single instruction. With fine-grained time-slicing, tasks would finish in the order of length, as with SJF, but a bit slower: a task A will take as long as it would have under SJF, plus a factor proportional to the number of other tasks as long as $A \times$ the length of A .

Unfortunately, Round Robin has some weaknesses. Figure 7.3 illustrates what happens for FIFO, SJF and Round Robin when several tasks start at roughly same time and are of the same length. Round Robin will rotate through the tasks, doing a bit of each, finishing them all at roughly the same time. This is nearly the worst possible scheduling policy for this workload! FIFO does much better, picking a task and sticking with it until it is done. Not only does FIFO reduce average response time for this workload relative to Round Robin, no task is worse off under FIFO — every task finishes at least as early as it would have under Round Robin. Time slicing added overhead without any benefit. Finally,

Figure 7.3: Completion times with FIFO, SJF, and Round Robin when scheduling equal length tasks.

Simultaneous Multi-Threading

Although zero overhead switching may seem far-fetched, most modern processors do a form of it called *simultaneous multi-threading (SMT)* or *hyperthreading*. With SMT, each processor simulates two (or more) virtual processors, alternating between them on a cycle by cycle basis. Since most tasks need to wait for memory from time to time, the other task can use the processor during those gaps, or vice versa. In normal operation, neither task is significantly slowed when running on an SMT.

You can test whether your computer implements SMT by testing how fast the processor operates when it has one or more tasks, each running a tight loop of arithmetic operations. (Note that on a multicore, you will need to create enough tasks to fill up each of the cores, or physical processors, before the system will begin to use SMT.) With one task per physical processor, each task will run at the maximum rate of the processor. With an SMT of two, and two tasks per processor, each task will run at somewhat less than the maximum rate, but each task will run at approximately the same uniform speed. As you increase the number of tasks beyond the SMT level, however, the operating system will begin to use coarse-grained time-slicing, so tasks will progress in spurts — alternating time on and off the processor.

Figure 7.4: Scheduling behavior with Round Robin when running a mixture of I/O-bound and compute-bound tasks.

consider what SJF does on this workload. SJF schedules tasks in exactly the same order as FIFO. The first task that arrives will be assigned the processor, and as soon as it executes a single instruction, it will have less time remaining than all of the other tasks, and so it will run to completion. Since we know SJF is optimal, this means that both FIFO and Round Robin are optimal for some workloads and pessimal for others, just different ones in each case.

Depending on the time quantum, Round Robin can also be quite poor when running a mixture of I/O-bound and compute-bound tasks. I/O-bound tasks often need very short periods on the processor in order to compute the next I/O operation to issue. Any delay to get scheduled onto the processor can lead to system-wide slowdowns. For example, in a text editor, it often takes only a few milliseconds to echo a keystroke to the screen, a delay much faster than human perception. However, if we are sharing the processor between a text editor and several other tasks using Round Robin, the editor must wait several time quanta to be scheduled for each keystroke — with a 100ms time quantum, this can become annoyingly apparent to the user.

Figure 7.4 illustrates similar behavior with a disk-bound task. Suppose we have a task that computes for 1ms and then uses the disk for 10ms, in a loop. Running alone, it can keep the disk almost completely busy. Suppose we also have two compute bound tasks; again, running by themselves, they can keep the processor busy. What happens when we run the disk-bound and compute-bound tasks at the same time? With Round Robin and a time quantum of

100ms, the disk-bound task slows down by nearly a factor of twenty — each time it needs the processor, it must wait nearly 200ms for its turn. SJF on this workload would perform well — prioritizing short tasks at the processor keeps the disk-bound task busy, while only modestly slowing down the compute-bound tasks.

If you have ever tried to surf the web while doing a large download, you can see that network operations visibly slow during the download. This is even though your browser may need to transfer only a very small amount of data to provide good responsiveness. The reason is quite similar. Browser packets get their turn, but only after being queued behind a much larger number of packets for the bulk download. Prioritizing the browser’s packets would have only a minimal impact on the download speed and a large impact on the perceived responsiveness of the system.

7.1.4 Max-Min Fairness

In many settings, a fair allocation of resources is as important to the design of a scheduler as responsiveness and low overhead. On a multi-user machine or on a server, we do not want to allow a single user to be able to monopolize the resources of the machine, degrading service for other users. While it might seem that fairness has little value in single-user machines, individual applications are often written by different companies, each with an interest in making their application performance look good even if that comes at a cost of degrading responsiveness for other applications. Further, some applications may run inside a single process, while others may create many processes, and each process may involve multiple threads. Round robin among threads can lead to starvation if applications with only a single thread are competing with applications with hundreds of threads.

We can be concerned with fair allocation at any of these levels of granularity: threads within a process, processes for a particular user, users sharing a physical machine. For simplicity, our discussion will assume we are interested in providing fairness among processes — the same principles apply if the unit receiving resources is the user, application or thread.

Fairness is easy if all processes are compute-bound: Round Robin will give each process an equal portion of the processor. In practice, however, different processes consume resources at different rates. An I/O-bound process may need only a small portion of the processor, while a compute-bound process is willing to consume all available processor time. What is a fair allocation when there is a diversity of needs?

One possible answer is to say that whatever Round Robin does is fair — after all, each process gets an equal chance at the processor. As we saw above, however, Round Robin can result in I/O-bound processes running at a much slower rate than they would if they had the processor to themselves, while

compute-bound processes are barely affected at all. That hardly seems fair!

While there are many possible definitions of fairness, a particularly useful one for operating system resource allocation is called *max-min fairness*. Max-min fairness iteratively maximizes the minimum allocation given to a particular process (user, application or thread) until all resources are assigned. If all processes are compute-bound, we maximize the minimum by giving each process exactly the same share of the processor.

Definition: max-min fairness

The behavior of max-min fairness is more interesting if some processes cannot use their entire share, for example, because they are short-running or are I/O-bound. If so, we give those processes their entire request and redistribute the unused portion to the remaining processes. Some of the processes receiving the extra portion may not be able to use all of their revised share, and so we must iterate, redistributing any unused portion. At some point, if the smallest remaining request cannot be fully satisfied, we divide the remainder equally among all remaining processes.

Consider the example in the previous section. The disk-bound process needed only 10% of the processor to keep busy, but Round Robin only gave it 0.5% of the processor, and each of the two compute-bound processes received nearly 50% of the processor. Max-min fairness would assign 10% of the processor to the I/O-bound processes, and split the remainder equally among the two compute-bound processes, with 45% each.

A hypothetical but completely impractical implementation of max-min would be to give the processor at each instant to whichever process has received the least portion of the processor. In the example above, the disk-bound task would always be instantly scheduled, preempting the compute-bound processes. However, we've already seen why this wouldn't work well. With two equally long tasks, as soon as we execute one instruction in one task, it would have received more resources than the other one, so to preserve "fairness" we would need to instantly switch to the next task.

We can approximate a max-min fair allocation by relaxing this constraint — to allow a process to get ahead of its fair allocation by one time quantum. Every time the scheduler needs to make a choice, it chooses the task for the process with the least accumulated time on the processor. If a new process arrives on the queue with much less accumulated time, such as the disk-bound task, it will preempt the process, but otherwise the current process will complete its quantum. Tasks may get up to one time quantum more than their fair share, but over the long term the allocation will even out.

The algorithm we just described was originally defined for network, and not processor, scheduling. If we share a link between a browser request and a long download, we will get reasonable responsiveness for the browser if we have approximately fair allocation — the browser needs few resources, and so its packets will always be scheduled ahead of the packets from the download.

Even this approximation, though, can be computationally expensive, since

it requires tasks to be maintained on a priority queue. For some server environments, there can be tens or even hundreds of thousands of scheduling decisions to be made every second. To reduce the computational overhead of the scheduler, most commercial operating systems use a somewhat different algorithm, to the same goal, which we describe next.

7.1.5 Case Study: Multi-level Feedback

Most commercial operating systems, including Windows, MacOS, and Linux, **Definition: Multi-level Feedback Queue (MFQ)** use a scheduling algorithm called *Multi-level Feedback Queue (MFQ)*. MFQ is designed to achieve several simultaneous goals:

- **Responsiveness.** Run short tasks quickly, as in SJF.
- **Low Overhead.** Minimize the number of preemptions, as in FIFO, and minimize the time spent making scheduling decisions.
- **Starvation-Freedom.** All tasks should make progress, as in Round Robin.
- **Background Tasks.** Defer system maintenance tasks, such as disk defragmentation, so they do not interfere with user work.
- **Fairness.** Assign (non-background) processes approximately their max-min fair share of the processor.

MFQ is an extension of Round Robin. Instead of only a single queue, MFQ has multiple Round Robin queues, each with a different priority level and time quantum. Tasks at a higher priority level preempt lower priority tasks, while tasks at the same level are scheduled in Round Robin fashion. Further, higher priority levels have *shorter* time quanta than lower levels.

Tasks are moved between priority levels to favor short tasks over long ones. A new task enters at the top priority level. Every time the task uses up its time quantum, it drops a level; every time the task yields the processor because it is waiting on I/O, it stays at the same level (or is bumped up a level); and if the task completes it leaves the system.

Figure 7.5 illustrates the operation of an MFQ with four levels. A new compute-bound task will start as high priority, but it will quickly exhaust its time-quantum and fall to the next lower priority, and then the next. Thus, an I/O-bound task needing only a modest amount of computing will almost always be scheduled quickly, keeping the disk busy. Compute-bound tasks run with a long time-quantum to minimize switching overhead while still sharing the processor.

So far, the algorithm we've described does not achieve starvation freedom or max-min fairness. If there are too many I/O-bound tasks, the compute-bound

Figure 7.5: Multi-level Feedback Queue when running a mixture of I/O-bound and compute-bound tasks.

tasks may receive no time on the processor. To combat this, the MFQ scheduler monitors every process to ensure it is receiving its fair share of the resources. At each level, Linux actually maintains two queues — tasks whose processes have already reached their fair share are only scheduled if all other processes at that level have also received their fair share. Periodically, any process receiving less than its fair share will have its tasks increased in priority; equally, tasks that receive more than their fair share can be reduced in priority.

Adjusting priority also addresses strategic behavior. From a purely selfish point of view, a task can attempt to keep its priority high by doing a short I/O request immediately before its time quantum expires. Eventually the system will detect this and reduce its priority to its fair-share level.

Our previously hapless supermarket manager reads a bit farther into the textbook and realizes that express lanes are a form of multi-level queue. By limiting express lanes to customers with a few items, the manager can ensure short tasks complete quickly, reducing average response time. The manager can also monitor wait times, adding extra lanes to ensure that everyone is served reasonably quickly.

7.1.6 Summary

We summarize the lessons from this section:

- FIFO minimizes overhead.
- If tasks are variable in size, then FIFO can have very poor average response time.
- FIFO is optimal in terms of average response time if tasks are equal size.
- Considering only the processor, SJF is optimal in terms of average response time.
- SJF is worst in terms of variance in response time.
- Round Robin approximates SJF if tasks have different sizes.
- Round Robin is the worst policy possible if tasks are of equal size.
- Tasks that intermix processor and I/O benefit from SJF and can do poorly under Round Robin.
- Max-min fairness can improve response time for I/O-bound tasks.
- By manipulating the assignment of tasks to priority queues, an MFQ scheduler can achieve responsiveness, low overhead, and fairness.

In the rest of this chapter, we extend these ideas to multiprocessors, real-time settings, energy-constrained environments, and overloaded conditions.

7.2 Multiprocessor scheduling

Today, most general-purpose computers are multiprocessors; even power-constrained smartphones often have two or more processors. Physical constraints in circuit design make it easier to add computational power by adding processors, or cores, onto a single chip, rather than making individual processors faster. Many high-end desktops and servers have multiple processing chips, each with multiple cores, and each core with hyperthreading. This trend is likely to accelerate, with systems of the future having dozens or perhaps hundreds of processors per computer.

This poses two questions for operating system scheduling:

- How do we make effective use of multiple cores for running sequential tasks?
- How do we adapt scheduling algorithms for parallel applications?

7.2.1 Scheduling Sequential Applications on Multiprocessors

Consider a server handling a very large number of web requests. A common software architecture for servers is to allocate a separate thread for each user connection. Each thread consults a shared data structure to see which portions of the requested data are cached, and fetches any missing elements from disk. The thread then spools the result out across the network.

How should the operating system schedule these server threads? Each thread is I/O-bound, repeatedly reading or writing data to disk and the network, and therefore makes many small trips through the processor. Some requests may require more computation; to keep average response time low, we will want to favor short tasks.

A simple approach would be to use a centralized multi-level feedback queue, with a lock to ensure only one processor at a time is reading or modifying the data structure. Each idle processor takes the next task off the MFQ and runs it. As the disk or network finishes requests, threads waiting on I/O are put back on the MFQ and executed by the network processor that becomes idle.

There are several potential performance problems with this approach:

- **Contention for the MFQ lock.** Depending on how much computation each thread does before blocking on I/O, the centralized lock may become a bottleneck, particularly as the number of processors increases.
- **Cache Coherence Overhead.** Although only a modest number of instructions are needed for each visit to the MFQ, each processor will need to fetch the current state of the MFQ from the cache of the previous processor to hold the lock. On a single processor, the scheduling data structure might be already loaded into the cache, but with a single copy of the scheduling data structure on a multiprocessor, the data is unlikely to be cached on the relevant processor. Fetching remote data can slow the speed of the processor down by two to three orders of magnitude. Since this occurs while holding the MFQ lock, that lock can become even more of a bottleneck.
- **Limited Cache Reuse.** Threads often run on a different processor each time they are scheduled, so that they must refetch any data they may have previously cached. Even on a uniprocessor, some of the thread's data will have been displaced from the cache during the time it was blocked, but on-chip caches are so large today that much of the thread's data may remain cached.

For these reasons, commercial operating systems such as Linux use a *per-processor* data structure: a separate copy of the multi-level feedback queue for

Figure 7.6: Per-processor scheduling data structures.

each processor. Figure ?? illustrates this. Each processor uses *affinity scheduling*: once a thread is scheduled on a processor, it is returned to the same processor when it is rescheduled, maximizing cache reuse. Each processor looks at its own copy of the queue for new work to do; this can mean that some processors can idle while others have work waiting to be done. Rebalancing occurs only if the queue lengths are persistent enough to compensate for the time to reload the cache for the migrated threads. Because rebalancing is possible, the per-processor data structures must still be protected by locks, but in the common case the next processor to use the data will be the last one to have written it, minimizing cache coherence overhead and lock contention.

Definition: **affinity scheduling**

7.2.2 Scheduling Parallel Applications

A different set of challenges occurs when scheduling parallel applications onto a multiprocessor. There is often a natural decomposition of a parallel application onto a set of processors. For example, an image processing application may divide the image up into equal size chunks, and assign one to each processor. While the application could divide the image into many more chunks than processors, this comes at a cost in efficiency: less cache reuse and more communication to coordinate work at the boundary between each chunk.

If there are multiple applications running at the same time, the application may receive fewer or more processors than it expected or started with. Applications can come and go, acquiring processing resources and releasing them. Even without multiple applications, the operating system itself will have system tasks to run from time to time, disrupting the mapping of parallel work onto a fixed number of processors.

Oblivious Scheduling

Definition: **oblivious scheduling**

One might imagine that the scheduling algorithms we've already discussed can take care of these cases. Each thread is time-sliced onto the available processors; if two or more applications create more threads in aggregate than processors, multi-level feedback will ensure that each thread makes progress and receives a fair share of the processor. This is often called *oblivious scheduling*, as the operating system scheduler operates without knowledge of the intent of the parallel application — each thread is scheduled as a completely independent entity.

Unfortunately, several problems can occur with oblivious scheduling on multiprocessors:

Figure 7.7: Bulk synchronous design pattern for a parallel program; preempting one processor can stall the synchronization after each step.

Figure 7.8: Producer-consumer design pattern for a parallel program; preempting one stage can stall the remainder.

- **Bulk synchronous delay.** A common design pattern in parallel programs is to split work into roughly equal sized chunks; once each chunk is done, the processors synchronize, waiting for every chunk to complete before communicating their results to the next stage of the computation. This *bulk synchronous* parallelism is easy to manage — each processor works independently, sharing its results only with the next stage in the computation. Google MapReduce is a widely used bulk synchronous application.

Definition: **bulk synchronous**

Figure 7.7 illustrates the problem with bulk synchronous computation under oblivious scheduling. At each step, the computation is limited by the slowest processor to complete that step. If a processor is preempted, its work will be delayed, stalling the remaining processors until the last one is scheduled. Even if one of the waiting processors picks up the preempted task, a single preemption can delay the entire computation by a factor of two, or possibly even more with cache effects. Since the application does not know that a processor was preempted, it cannot adapt its decomposition for the available number of processors, so each step is similarly delayed until the processor is returned.

- **Producer-consumer delay.** Some parallel applications use a producer-consumer design pattern, where the results of one thread are fed to the next thread, and the output of that thread is fed onward, as in Figure 7.8. Preempting a thread in the middle of a producer-consumer chain can stall all of the processors in the chain.

- **Critical path delay.** More general parallel programs have a *critical path* — the minimum sequence of steps for the application to compute its result, illustrated in Figure 7.9. Work off the critical path can be done in parallel, but its precise scheduling is less important. Preempting a thread on the critical path, however, will slow down the end result. Although the application programmer may know which parts of the computation are on the critical path, with oblivious scheduling, the operating system will not; it will be equally likely to preempt a thread on the critical path as off.

Definition: **critical path**

- **Preemption of lock holder.** Many parallel programs use locks and condition variables for synchronizing their parallel execution. Often, to reduce the cost of acquiring locks, parallel programs will use a “spin-then-wait” strategy — if a lock is busy, the waiting thread spin-waits briefly for

Figure 7.9: Critical path of a parallel program; delays on the critical path increase execution time.

Figure 7.10: Performance as a function of the number of processors, for some typical parallel applications. Some applications scale linearly with the number of processors; others achieve diminishing returns.

it to be released, and if the lock is still busy, it blocks and looks for other work to do. This can reduce overhead in the common case that the lock is held for only short periods of time. With oblivious scheduling, however, the lock holder can be preempted — other tasks will spin-then-wait until the lock holder is re-scheduled.

- **I/O.** Many parallel applications do I/O, and this can also cause problems if the operating system scheduler is oblivious to the application decomposition into parallel work. If a read or write request blocks in the kernel, the thread blocks as well. To reuse the processor while the thread is waiting, the application processor must have created more threads than processors, so that it can choose one to run in place of the blocked thread. However, if the thread doesn't block, that means that the scheduler has more threads than processors, and so needs to do time slicing to multiplex threads onto processors — causing all of the problems we have listed above.

Co-scheduling

One possible approach to some of these issues is to schedule all of the tasks of a program together. This is called *co-scheduling*. The application picks some decomposition of work into some number of threads, and those threads either run together or not at all. If the operating system needs to schedule a different application, if there are insufficient idle resources, it preempts all of the processors of an application to make room.

Because of the value of co-scheduling, commercial operating systems like Linux, Windows, and MacOS have mechanisms that can be used to co-schedule a single application. This is appropriate when a server is dedicated to a single use, such as a database needing precise control over thread assignment. The application can *pin* each thread to a specific processor and (with the appropriate permissions) mark it to run with high priority. A small subset of the system's processors is reserved to run other applications and system management tasks, multiplexed in the normal way but without interfering with the primary application.

For multiplexing multiple parallel applications, however, co-scheduling can be inefficient. Figure 7.10 illustrates why. It shows the typical performance

of parallel programs as a function of the number of processors. While some applications have perfect speedup and can make efficient use of many processors, other applications reach a point of diminishing returns, and still others have a maximum parallelism.

An implication of Figure 7.10 is that it is more usually more efficient to run two parallel programs each with half the number of processors, than to time slice the two programs, each co-scheduled onto all of the processors. Since the number of available processors is a dynamic property in a multiprogrammed setting, how does the application know how many processors to use?

Scheduler activations

A solution, recently added to Windows, is to make the assignment and re-assignment of processors to applications visible to applications. Applications are given an execution context, or *scheduler activation*, on each processor assigned to the application; the application is informed explicitly, via an upcall, whenever a processor is added to its allocation or taken away. Blocking on an I/O request also causes an upcall to allow the application to repurpose the processor while the thread is waiting for I/O.

Definition: **scheduler activation**

As we noted in an earlier chapter, user-level thread management is possible with scheduler activations. The operating system kernel assigns processors to applications, either evenly or according to some priority weighting. Each application then schedules its user-level threads onto the processors assigned to it, changing its allocation as the number of processors varies due to external events. If no other application is running, an application can use all of the processors of the machine; with more contention, the application must remap its work onto a smaller number of processors.

7.3 Energy-aware scheduling

Naively, have a certain amount of computation to do, not much difference if we are battery constrained or not. We can put off background tasks, but its likely that's not all that significant.

Turns out that modern architectures have a number of ways of trading performance for power consumption.

Slow down CPU to make it more efficient. Turn off processors that are not being used, or not being used much Disable memory that is not being used.

How do we know if these are worthwhile things to do?

Response time as a function of resources – diminishing returns

Value as a function of response time – S curve

Giant optimization, but

7.4 Real-time scheduling

Meet deadlines by doing edf, but how do we keep from missing deadlines?

control theory: adjust processing rate to ensure meet goals, but if there's a delay between adjusting the rate and seeing the change in your metric, then can get an unstable system.

7.5 Queuing theory

7.5.1 Load v. response time

queuing delay – response time increases when load increases ; suppose system takes 100ms to process each request suppose that requests arrive perfectly evenly spaced what is response time (DRAW GRAPH) 1/sec 2/sec (every 500ms) 3/sec (every 333 1/3 ms) ... 9/sec 9.99999/sec (every 100.001ms) 10.0001/sec (every 99.999ms) 10/sec (every 100ms exactly)

suppose requests arrive in bursts at the start of each second. What is response time (DRAW GRAPH) 1/sec 2/sec ...

suppose requests arrive "randomly" – a bunch of different users sending requests s.t., receiving a request from user A doesn't affect the expected time to next request from some other user – exponential distribution (define); memoryless

memoryless is an approximation, a model, that allows us to abstract away time markov graph – we can calculate precisely how often system is in various states these graphs allow us to calculate the impact of scheduling choices (such as RR vs. FIFO) on response time and throughput

if arrivals/service times are random, scheduling is FIFO, and arrivals don't depend on wait times, graph looks like this: $t = 1/(1-\text{utilization})$

notice it is between "perfectly even" arrivals and "perfectly bursty" arrivals

what happens to queue length as we increase load? under perfectly even, zero under perfectly bursty, size of burst under random, grows unbounded with utilization

what happen to throughput as we increase load? ditto

Some more corner cases:

what if we increase load beyond what the system can handle? If arrivals are independent of response time, then system will get infinite queues – its not a steady state.

Model vs. Reality

One way of answering what happens under different load conditions, is to simulate the different loads and so you can see exactly how response time varies. For the precise answer, it matters exactly when each job arrives. But we also want to understand the behavior of a system under a range of conditions, and for that we construct a model, an approximation of reality.

Models are not true or false. If by abstracting detail, we can still be approximately correct, then the model can be useful. After all, systems get used in ways that are quite different from how we predict they might!

Picking a workload

anecdote: students were glum because they thought a system they built had horrible performance (e.g., 700ms response time). It turned out the system was fine, but their clients were overloading it)

what if rate of arrivals depends on response time? Eg., fixed number of customers; each one can't ask for more unless you respond. That means you can increase load infinitely, and you still be in steady state.

what if there are multiple servers? Is it better to have one queue for everyone, or one queue per server, or does it matter? Issue is that if a server can be idle while others have work to do, then that is effectively like reducing system capacity, so that increases response time.

what if we double the speed of a CPU that's 90unfortunately, you can't say: suppose each job needs to do both CPU and I/O. If I/O was 80

lessons:

- load increases response time
- burstiness increases response time
- overload is catastrophic – design systems to operate with low to moderate load (if load hits 99– have an overload control plan)
- scheduling often doesn't matter
- measuring systems – know your load – measure 1 request at a time – min latency – increase load to saturate system – max throughput Now

you have a pretty good idea of where your system will be under range of loads

- Know your arrival pattern mathematically convenient to assume "exponential distribution" but...

7.5.2 Overload management

load limiting v. scheduling when load is low – nothing matters when load is medium – scheduling may matter when overload – load limiting matters

many servers do fifo request scheduling + overload management highly variable load – more sophisticated scheduling not that important – provision the system so load is usually low – during normal operation, scheduling doesn't matter much – overload management to avoid falling over under unexpected load

overload management – need to do less work when overloaded e.g., reject some requests (e.g., movie stream server rejects requests to start new movies so that it can continue to provide good streaming to streams that have already started. Alternative to giving a few customers good service and others bad (no) service is to give all customers bad service.

e.g., do less work (e.g., movie stream server reduces video quality/bit rate for everyone; e.g., ebay updates auction listings every 10 minutes instead of every minute)

dynamics can also work the other way as well: if amount of work per task increases as the load increases, then response times can soar even faster, and throughput decreases as we add load. This makes overload management even more important.

examples: highway traffic, with onramp limiters time-slicing in the presence of caches (more time slices, fewer cache hits) network traffic, with packet losses as network becomes loaded – keep from sending data into the network another example: virtual memory

overload management gets less attention than scheduling because messier. Scheduling is clean policy choice. We build thread systems that work no matter what scheduler you choose. You can change scheduler and not affect rest of system

in contrast, overload management is visible to users; may affect other parts of system design

Ebay keynote at laddis – under overload, you need to turn something off, the only question is, do you choose, or let it choose you ebay's scaling odyssey (cs.cornell.edu)

CNN going to a static page on 9/11

amazon returns a result to user, even if it isn't ready yet (apologies later)
ncaa broadcaster turns people away, rather than giving everyone poor service
(analogy: restaurant and instead of a having a line, you let everyone sit down at a table)

7.6 Case Study: data center servers

- Customers get assigned to a node when they arrive at the web site, to the node with the least load
- Additional requests from that user get assigned to the same node (affinity scheduling)
- Prevent users from hogging resources; favor short tasks over long ones or bin requests based on source IP
- want to ensure response time stays small, so add resources if load increases
- it can take time to bring online new resources, so need to model with control theory

Part III

Memory Management

Chapter 8

Address Translation

Language forces us to perceive the world as man presents it to us. – Julia Penelope

*Words, like Nature, half reveal
And half conceal the Soul within.* – Alfred, Lord Tennyson

Address translation has many benefits (it is not just about virtual memory).

- **Protection.** Keep each process isolated
- **Sharing.** Allow memory to be shared between processes (code segments, but also shared regions of memory, copy on write). Memory sharing is important not only because we might run out of memory, but more importantly to make caches more efficient.
- **Virtualization.** Provide OS services transparently to applications: the illusion of infinite memory, but also virtual machines, process checkpointing and migration, information flow control, and other applications.

This chapter has three main ideas:

- Address translation concept
- How to make address translation flexible
- How to make address translation efficient

8.1 Address translation concept

linking and base and bounds

8.2 Segmentation and Paging

8.2.1 Segmentation

8.2.2 Paging

8.2.3 Multi-level paging

8.2.4 Multilevel segmentation and paging

e.g., the x86

8.3 Efficient address translation

8.3.1 Translation lookaside buffers

Software managed TLBs

8.3.2 Virtually addressed caches

8.3.3 Consistency management

I/O, core map, TLB and page shootdown

tagged TLBs and virtual caches

8.4 Software address translation

8.4.1 Software fault isolation

e.g., native client

8.4.2 Dynamic translation

Java and android virtual machines

8.5 Conclusions and future directions

Chapter 9

Caching and Virtual Memory

Cash is king – Per Gyllenhammar

Caching is central to many aspects of computer systems: not just memory, but also file systems, naming, networking, servers, etc.

Can cache data, or the results of computation. Some questions:

- When is caching useful? (e.g., if cost of regenerating the data is more than the cost of storing it until it is next reused)
- How do we know if cache is valid?
- How do we choose what to keep in the cache?

In this chapter, we're going to focus on caching disk contents, but the underlying principles apply elsewhere.

Programmer can manage the transfer to/from disk explicitly, but may find it easier not to have to do so. E.g., memory mapped files are widely used – data in the file is accessed as if the file is stored in memory, but the actual file is on disk. Virtual memory is another example.

9.1 Cache concept: when it works and when it doesn't

when caching works: working set model

when caching doesn't work (as well as you might like): Zipf – sequential access to large data

9.2 Hardware cache management

Direct mapped

Set associative

Page coloring

9.3 Memory mapped files and virtual memory

9.3.1 Memory allocation policies

FIFO, LRU, LFU, optimal, self-paging

9.3.2 Implementing memory allocation

Emulating modify and use bits

9.4 Conclusions and future directions

Chapter 10

Applications of Memory Management

All problems in computer science can be solved by another level of indirection.

– David Wheeler

Advanced applications of address translation hardware.

10.1 Zero copy input/output

10.2 Copy on write

10.3 Process checkpointing

10.4 Recoverable memory

10.5 Information flow control

10.6 External pagers

databases and garbage collection

10.7 Virtual machine address translation

10.8 Conclusions and future directions

Part IV

Persistent Storage

Chapter 11

File Systems: Introduction and Overview

Memory is the treasury and guardian of all things.

— Marcus Tullius Cicero

Computers must be able to reliably store data. Individuals store family photos, music files, and email folders; programmers store design documents and source files; office workers store spreadsheets, text documents, and presentation slides; and businesses store inventory, orders, and billing records. In fact, for a computer to work at all, it needs to be able to store programs to run and the operating system, itself.

For all of these cases, users demand a lot from their storage systems:

- **Reliability.** A user's data should be safely stored even if a machine's power is turned off or its operating system crashes. In fact, much of this data is so important that users expect and need the data to survive even if the devices used to store it are damaged. For example, many modern storage systems continue to work even if one of the magnetic disks storing the data malfunctions or even if a data center housing some of the system's servers burns down!
- **Large capacity and low cost.** Users and companies store enormous amount of data, so they want to be able to buy high capacity storage for a low cost. For example, it takes about 350 MB to store an hour of CD-quality losslessly encoded music, 4 GB to store an hour-long high-definition home video, and about 1 GB to store 300 digital photos. As a

Unit	Binary Bytes	Decimal Bytes
MB	2^{20}	10^6
GB	2^{30}	10^9
TB	2^{40}	10^{12}
PB	2^{50}	10^{15}
EB	2^{60}	10^{18}

result of these needs, many individuals own 1 TB or more of storage for their personal files. This is an enormous amount: if you printed 1 TB of data as text on paper, you would produce a stack about 20 miles high. In contrast, for less than \$100 you can buy 1 TB of storage that fits in a shoebox.

- **High performance.** For programs to use data, they must be able to access it, and for programs to use large amounts of data, this access must be fast. For example, users want program start-up to be nearly instantaneous, a business may need to process hundreds or thousands of orders per second, or a server may need to stream a large number of video files to different users.
- **Named data.** Because users store a large amount of data, because some data must last longer than the process that creates it, and because data must be shared across programs, storage systems must provide ways to easily identify data of interest. For example, if you can name a file (e.g., `/home/alice/assignments/hw1.txt`) you can find the data you want out of the millions of blocks on your disk, you can still find it after you shut down your text editor, and you can use your email program to send the data produced by the text editor to another user.
- **Controlled sharing.** Users need to be able to share stored data, but this sharing needs to be controlled. As one example, you may want to create a design document that everyone in your group can read and write, that people in your department can read but not write, and that people outside of your department cannot access at all. As another example, it is useful for a system to be able to allow anyone to execute a program while only allowing the system administrator to change the program.

Nonvolatile storage and file systems. The contents of a system’s main DRAM memory can be lost if there is an operating system crash or power failure.

Definition: **nonvolatile storage**

Definition: **persistent storage**

Definition: **stable storage**

In contrast, *nonvolatile storage* is durable and retains its state across crashes and power outages; nonvolatile storage is also called or *persistent storage* or *stable storage*. Nonvolatile storage can also have much higher capacity and lower cost than the volatile DRAM that forms the bulk of most system’s “main memory.”

However, nonvolatile storage technologies have their own limitations. For example, current nonvolatile storage technologies such as magnetic disks and high-density flash storage do not allow random access to individual words of storage; instead, access must be done in more coarse-grained units—512, 2048, or more bytes at a time.

Furthermore, these accesses can be much slower than access to DRAM; for example, reading a sector from a magnetic disk may require activating a motor to move a disk arm to a desired track on disk and then waiting for the spinning disk to bring the desired data under the disk head. Because disk accesses

Goal	Physical Characteristic	Design Implication
High performance	Large cost to initiate IO access	Organize data placement with <i>files</i> , <i>directories</i> , <i>free space bitmap</i> , and <i>placement heuristics</i> so that storage is accessed in large sequential units <i>Caching</i> to avoid accessing persistent storage
Named data	Storage has large capacity, survives crashes, and is shared across programs	Support <i>files</i> and <i>directories</i> with meaningful names
Controlled sharing	Device stores many users' data	Include access-control <i>metadata</i> with files
Reliable storage	Crash can occur during update Storage devices can fail Flash memory cells can wear out	Use <i>transactions</i> to make a set of updates atomic Use <i>redundancy</i> to detect and correct failures Move data to different storage locations to even the wear

Figure 11.1: Characteristics of persistent storage devices affect the design of an operating system's storage abstractions.

involve motors and physical motion, the time to access a random sector on a disk can be around 10 milliseconds. In contrast, DRAM latencies are typically under 100 nanoseconds. This large difference—about five orders of magnitude in the case of spinning disks—drives the operating system to organize and use persistent storage devices differently than main memory.

File systems are a common operating system abstraction to allow applications to access nonvolatile storage. File systems use a number of techniques to cope with the physical limitations of nonvolatile storage devices and to provide better abstractions to users. For example, Figure 11.1 summarizes how physical characteristics motivate several key aspects of file system design.

- **Performance.** File systems amortize the cost of initiating expensive operations—such as moving a disk arm or erasing a block of solid state memory—by grouping where its placement of data so that such operations access large, sequential ranges of storage.
- **Naming.** File systems group related data together into directories and files and provide human-readable names for them (e.g., `/home/alice/Pictures/summer-vacation/hiking.jpg`.) These names for data remain meaningful even after the program that creates the data exits, they help

users organize large amounts of storage, and they make it easy for users to use different different programs to create, read, and edit, their data.

- **Controlled sharing.** File systems include metadata about who owns which files and which other users are allowed to read, write, or execute data and program files.
- **Reliability.** File systems use transactions to atomically update multiple blocks of persistent storage, similar to how the operating system uses critical sections atomically update different data structures in memory.

To further improve reliability, file systems store checksums with data to detect corrupted blocks, and they replicate data across multiple storage devices to recover from hardware failures.

Impact on application writers. Understanding the reliability and performance properties of storage hardware and file systems is important even if you are not designing a file system from scratch. Because of the fundamental limitations of existing storage devices, the higher-level illusions of reliability and performance provided by the file system are imperfect. An application programmer needs to understand these limitations to avoid having inconsistent data stored on disk or having a program run orders of magnitude slower than expected.

For example, suppose you edit a large document with many embedded images and that your word processor periodically auto-saves the document so that you would not lose too many edits if the machine crashes. If the application uses the file system in a straightforward way, several of unexpected things may happen.

- **Poor performance.** First, although file systems allow existing bytes in a file to be overwritten with new values, they do not allow new bytes to be inserted into the middle of existing bytes. So, even a small update to the file may require rewriting the entire file either from beginning to end or at least from the point of the first insertion to the end. For a multi-megabyte file, each auto-save may end up taking as much as a second.
- **Corrupt file.** Second, if the application simply overwrites the existing file with updated data, an untimely crash can leave the file in an inconsistent state, containing a mishmash of the old and new versions. For example, if a section is cut from one location and pasted in another, after a crash the saved document may end up with copies of the section in both locations, one location, or neither location; or it may end up with a region that is a mix of the old and new text.
- **Lost file.** Third, if instead of overwriting the document file, the application writes updates to a new file, then deletes the original file, and finally

moves the new file to the original file's location, an untimely crash can leave the system with no copies of the document at all.

Programs use a range of techniques to deal with these types of issues. For example, some structure their code to take advantage of the detailed semantics of some operating systems. For example, some operating systems guarantee that when a file is renamed and a file with the target name already exists, the target name will always refer to either the old or new file, even after a crash in the middle of the rename operation. In such a case, an implementation can create a new file with the new version of the data and use the rename command to atomically replace the old version with the new one.

Other programs essentially build a miniature file system over the top of the underlying one, structuring their data so that the underlying file system can better meet their performance and reliability requirements.

For example, a word processor might use a sophisticated document format, allowing it to, for example, add and remove embedded images and to always update a document by appending updates to the end of the file.

As another example, a data analysis program might improve its performance by organizing its accesses to input files in a way that ensures that each input file is read only once and that it is read sequentially from its start to its end.

Or, a browser with a 1 GB on-disk cache might create 100 files, each containing 10 MB of data, and group a given web site's objects in a sequential region of a randomly selected file. To do this, the browser would need to keep metadata that maps each cached web site to a region of a file, it would need to keep track of what regions of each file are used and which are free free, it would need to decide where to place a new web site's objects, and it would need to have a strategy for growing or moving a web site's objects as additional objects are fetched.

Roadmap. To get good performance and acceptable reliability, both application writers and operating systems designers must understand how storage devices and file systems work. This chapter and the next three discuss the key issues:

- **API and abstractions.** The rest of this chapter introduces file systems by describing a typical API and set of abstractions, and it provides an overview of the software layers that provide these abstractions.
- **Storage devices.** The characteristics of persistent storage devices strongly influence the design of storage system abstractions and higher level applications. Chapter 12 therefore explores the physical characteristics of common storage devices.

- **Implementing files and directories.** Chapter 13 describes how file systems keep track of data by describing several widely-used approaches to implementing files and directories.
- **Reliable storage.** Although we would like storage to be perfectly reliable, physical devices fall short of that ideal. Chapter 14 describes how storage systems use transactional updates and redundancy to improve reliability.

11.1 The file system abstraction

Today, almost anyone who uses a computer is familiar with the high-level file system abstraction. File systems provide a way for users to organize their data and to store it for long periods of time. For example, Bob's computer might store a collection of applications such as `/Applications/Calculator` and `/ProgramFiles/TextEdit` and a collection of data files such as `/home/Bob/correspondence/letter-to-mom.txt`, and `/home/Bob/Classes/OS/hw1.txt`.

Definition: **file system**

More precisely, a *file system* is an operating system abstraction that provides

Definition: **persistent data**

persistent, named data. *Persistent data* is stored until it is explicitly deleted,

Definition: **named data**

even if the computer storing it crashes or loses power. *Named data* can be accessed via a human-readable identifier that the file system associates with the file. Having a name allows a file to be accessed even after the program that created it has exited, and allows it to be shared by multiple applications.

There are two key parts to the file system abstraction: files, which define sets of data, and directories, which define names for files.

Definition: **file**

File. A *file* is a named collection of data in a file system. For example, the programs `/Applications/Calculator` or `/ProgramFiles/TextEdit` are each files, as are the data `/home/Bob/correspondence/letter-to-mom.txt` or `/home/Bob/Classes/OS/hw1.txt`.

Files provide a higher-level abstraction than the underlying storage device: they let a single, meaningful name refer to an (almost) arbitrarily-sized amount of data. For example `/home/Bob/Classes/OS/hw1.txt` might be stored on disk in blocks `0xA713F28`, `0xB3CA349A`, and `0x33A229B8`, but it is much more convenient to refer to the data by its name than by this list of disk addresses.

Definition: **file metadata**

A file's information has two parts, metadata and data. A *file's metadata* is information about the file that is understood and managed by the operating system. For example, a file's metadata typically includes the file's *size*, its *modification time*, its *owner*, and its *security information* such as whether it may be read, written, or executed by the owner or by other users.

Definition: **file data**

A *file's data* can be whatever information a user or application puts in it.

From the point of view of the file system, a file's data is just an array of untyped bytes. Applications can use these bytes to store whatever information they want in whatever format they choose. Some data have a simple structure. For example, an ASCII text file contains a sequence of bytes that are interpreted as letters in the English alphabet. Conversely, data structures stored by applications can be arbitrarily complex. For example, a .doc files can contain text, formatting information, and embedded objects and images, an ELF (Executable and Linkable File) files can contain compiled objects and executable code, or a database file can contain the information and indices managed by a relational database.

Directory. Whereas a file contains system-defined metadata and arbitrary data, directories provide names for files. In particular, a *directory* is a list of human-readable names and a mapping from each name to a specific underlying file or directory. One common metaphor is that a directory is a folder that contains documents (files) and other folders (directories).

Definition: **directory**

As Figure 11.2 illustrates, because directories can include names of other directories, they can be organized in a hierarchy so that different sets of associated files can be grouped in different directories. So, the directory `/bin` may include binary applications for your machine while `/home/tom` (Tom's "home directory") might include Tom's files. If Tom has many files, Tom's home directory may include additional directories to group them (e.g., `/home/tom/Music` and `/home/tom/Work`.) Each of these directories may have subdirectories (e.g., `/home/tom/Work/Class` and `/home/tom/Work/Docs`) and so on.

The string that identifies a file or directory (e.g., `/home/tom/Work/Class/OS/hw1.txt` or `/home/tom`) is called a *path*. Here, the symbol `/` (pronounced *slash*) separates components of the path, and each component represents an entry in a directory. So, `hw1.txt` is a file in the directory `OS`; `OS` is a directory in the directory `Work`; and so on.

Definition: **path**

If you think of the directory as a tree, then the root of the tree is a directory called, naturally enough, the *root directory*. Path names such as `/bin/ls` that begin with `/` define *absolute paths* that are interpreted relative to the root directory. So, `/home` refers to the directory called `home` in the root directory.

Definition: **root directory**

Definition: **absolute path**

Path names such as `Work/Class/OS` that do not begin with `/` define *relative paths* that are interpreted by the operating system relative to a process's *current working directory*. So, if a process's current working directory is `/home/tom`, then the relative path `Work/Class/OS` is equivalent to the absolute path `/home/tom/Work/Class/OS`.

Definition: **relative path**

Definition: **current**

Definition: **working directory**

When you log in, your shell's current working directory is set to your *home directory*. Processes can change their current working directory with the `chdir(path)` system call. So, for example, if you log in and then type `cd Work/Class/OS`, your current working directory is changed from your home directory to the subdirectory `Work/Class/OS` in your home directory.

Definition: **home directory**

Executing “untyped” files

Usually, an operating system treats a file’s data as an array of untyped bytes, leaving it up to applications to interpret a file’s contents. Occasionally, however, the operating system needs to be able to parse a file’s data.

For example, Linux supports a number of different executable file types such as the ELF and a.out binary files and tsch, csh, and perl scripts. You can run any of these files from the command line or using the `exec()` system call. E.g.,

```
> a.out
Hello world from hello.c compiled by gcc!
> hello.pl
Hello world from hello.pl, a perl script!
> echo "Hello world from /bin/echo, a binary supplied with your system!"
Hello world from /bin/echo, a binary supplied with your system!
```

To execute a file, the operating system must determine whether it is a binary file or a script. If it is the former, the operating system must parse the file to determine where in the target process’s memory to load code and data from the file and which instruction to start with. If it is the latter, the operating system must determine which interpreter program it should launch to execute the script.

Linux does this by having executable files begin with a *magic number* that identifies the file’s format. For example, ELF binary executables begin with the four bytes 0x7f, 0x45, 0x4c, and 0x46 (the ASCII characters DEL, E, L, and F); once an executable is known to be an ELF file, the ELF standard defines how the operating system should parse the rest of the file to extract and load the program’s code and data. Similarly, script files begin with #! followed by the name of the interpreter that should be used to run the script (e.g., a script might begin with #!/bin/sh to be executed using the Bourne shell or #!/usr/bin/perl to be executed using the perl interpreter).

Alternative approaches include determining a file’s type by its name *extension*—the characters after the last dot (.) in the file’s name (e.g., .exe, .pl, or .sh)—or including information about a file’s type in its metadata.

Multiple data streams

For traditional files, the file's data is a single logical sequence of bytes, and each byte can be identified by its offset from the start of the sequence (e.g., byte 0, byte 999, or byte 12481921 of a file.)

Some file systems support multiple sequences of bytes per file. For example, Apple's MacOS Extended file system supports multiple *forks* per file—a data fork for the file's basic data, a resource fork for storing additional attributes for the file, and multiple named forks for application-defined data. Similarly, Microsoft's NTFS supports *alternate data streams* that are similar to MacOS's named forks.

In these systems, when you open a file to read or write its data, you specify not only the file but also the fork or stream you want.

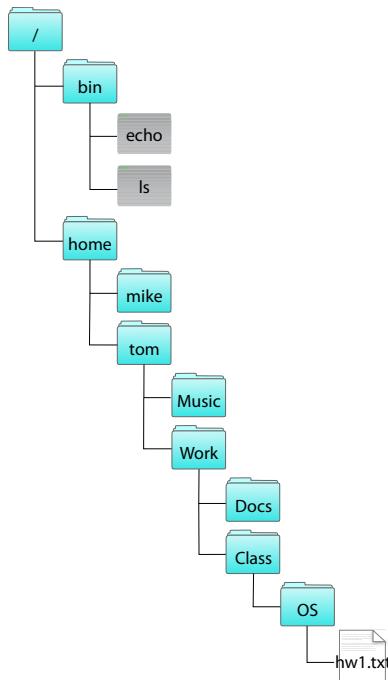


Figure 11.2: Example of a hierarchical organization of files using directories.

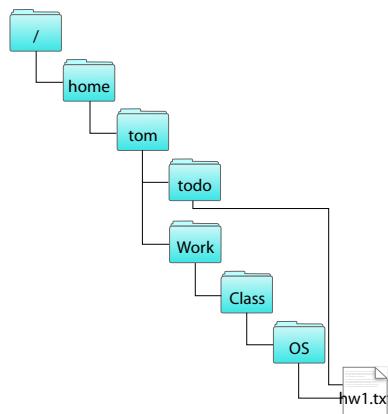


Figure 11.3: Example of a directed acyclic graph directory organization with multiple hard links to a file.

. and .. and ~

You may sometimes see path names in which directories are named ., .., or ~. E.g.,

```
> cd ~/Work/Class/OS
> cd ..
> ./a.out
```

., .., and ~ are special directory names in Unix. . refers to the *current working directory*, .. refers to the *parent of the current working directory*, and ~ refers to the *current user's home directory*.

So, the first shell command changes the current working directory to be the Work/Class/OS directory in the user's home directory (e.g., /home/tom/Work/Class/OS). The second command changes the current working directory to be the Work/Class directory in the user's home directory (e.g., ~/Work/Class or /home/tom/Work/Class.) The third command executes the program a.out from the current working directory (e.g., ~/Work/Class/a.out or /home/tom/Work/Class/a.out.)

If each file or directory is identified by exactly one path, then the directory hierarchy forms a tree. Occasionally, it is useful to have several different names for the same file or directory. For example, if you are actively working on a project, you might find it convenient to have the project appear in both your “todo” directory and a more permanent location (e.g., `/home/tom/todo/hw1.txt` and `/home/tom/Work/Class/OS/hw1.txt` as illustrated in Figure 11.3.)

The mapping between a name and the underlying file is called a *hard link*. Definition: **hard link** If a system allows multiple hard links to the same file, then the directory hierarchy may no longer be a tree. Most file systems that allow multiple hard links to a file restrict these links to avoid cycles, ensuring that their directory structures form a directed acyclic graph (DAG.) Avoiding cycles can simplify management by, for example, ensuring that recursive traversals of a directory structure terminate or by making it straightforward to use reference counting to garbage collect a file when the last link to it is removed.

In addition to hard links, many systems provide other ways to use multiple names to refer to the same file. See the sidebar for a comparison of hard links, soft links, symbolic links, shortcuts, and aliases.

Volume. Each instance of a file system manages files and directories for a volume. A *volume* is a collection of physical storage resources that form a logical storage device. Definition: **volume**

A volume is an abstraction that corresponds to a logical disk. In the simplest case, a volume corresponds to a single physical disk drive. Alternatively, a single physical disk can be partitioned and store multiple volumes or several physical disks can be combined so that a single volume spans multiple physical disks.

A single computer can make use of multiple file systems stored on multiple volumes by mounting multiple volumes in a single logical hierarchy. *Mounting* a volume on an existing file system creates a mapping from some path in the existing file system to the root directory of the mounted volume’s file system and lets the mounted file system control mappings for all extensions of that path. Definition: **mount**

For example, suppose a USB drive contains a file system with the directories `/Movies` and `/Backup` as shown in Figure 11.4. If Alice plugs that drive into her laptop, the laptop’s operating system might mount the USB volume’s file system with the path `/Volumes/usb1/` as shown in Figure 11.5. Then, if Alice calls `open(''/Volumes/usb1/Movies/vacation.mov'')`, she will open the file `/Movies/vacation.mov` from the file system on the USB drive’s volume. If, instead, Bob plugs that drive into his laptop, the laptop’s operating system might mount the volume’s file system with the path `/media/disk-1`, and Bob would access the same file using the path `/media/disk-1/Movies/vacation.mov`.

Hard links, soft links, symbolic links, shortcuts, and aliases

A hard link is a directory mapping from a file name directly to an underlying file. As we will see in Chapter 13, directories will be implemented by storing mappings from *file names* to *file numbers* that uniquely identify each file. When you first create a file (e.g., `/a/b`), the directory entry you create is a hard link to the new file. If you then use `link()` to add another hard link to the file (e.g., `link('/a/b', '/c/d')`), then both names are equally valid, independent names for the same underlying file. You could, for example, `unlink('/a/b')`, and `/c/d` would remain a valid name for the file.

Many systems also support *symbolic links* also known as *soft links*. A symbolic link is a directory mapping from a file name to *another file name*. If a file is opened via a symbolic link, the file system first translates the name in the symbolic link to the target name and then uses the target name to open the file. So, if you create `/a/b`, create a symbolic link from `/c/d/` to `/a/b`, and then `unlink /a/b`, the file is no longer accessible and `open('/c/d/')` will fail.

Although the potential for such dangling links is a disadvantage, symbolic links have a number of advantages over hard links. First, systems usually allow symbolic links to directories, not just regular files. Second, a symbolic link can refer to a file stored in a different file system or volume.

Some operating systems such as Microsoft Windows also support *shortcuts*, which appear similar to symbolic links but which are interpreted by the windowing system rather than by the file system. From the file system's point of view, a shortcut is just a regular file. The windowing system, however, treats shortcut files specially: when the shortcut file is selected via the windowing system, the windowing system opens that file, identifies the target file referenced by the shortcut, and acts as if the target file had been selected.

A MacOS file *alias* is similar to a symbolic link but with an added feature: if the target file is moved to have a new path name, the alias can still be used to reference the file.

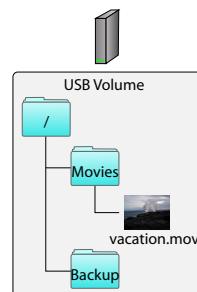


Figure 11.4: This USB disk holds a volume that is the physical storage for a file system.

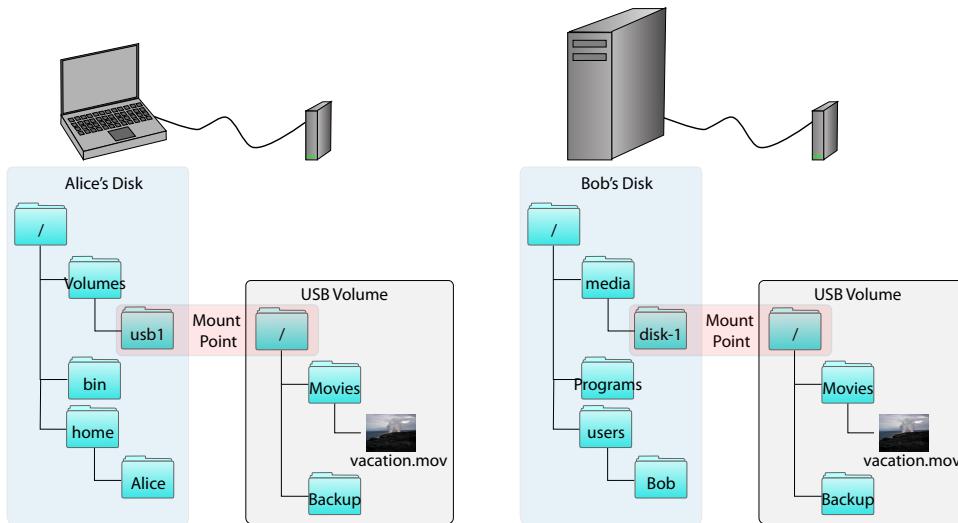


Figure 11.5: A volume can be mounted to another file system to join their directory hierarchies. For example, when the USB drive is connected to Alice’s computer, she can access the `vacation.mov` movie using the path `/Volumes/usb1/Movies/vacation.mov`, and when the drive is connected to Bob’s computer, he can access the movie using the path `/media/disk-1/Movies/vacation.mov`.

11.2 API

For concreteness, Figure 11.6 shows a simple file system API for accessing files and directories.

Creating and deleting files. Processes create and destroy files with `create()` and `unlink()`. `Create()` does two things: it creates a new file that has initial metadata but no other data, and it creates a name for that file in a directory.

`Link()` creates a hard link—a new path name for an existing file. After a successful call to `link()`, there are multiple path names that refer to the same underlying file.

`Unlink()` removes a name for a file from its directory. If a file has multiple names or links, `unlink()` only removes the specified name, leaving the file accessible via other names. If the specified name is the last (or only) link to a file, then `unlink()` also deletes the underlying file and frees its resources.

`Mkdir()` and `rmdir()` create and delete directories.

Example: Linking to files v. linking to directories.

Question: Systems such as Linux support a `link()` system call, but they do not allow new hard links to be created to a directory. E.g.,

Creating and deleting files	
<code>create(pathName)</code>	Create a new file with the specified name
<code>link(existingName, newName)</code>	Create a hard link—a new path name that refers to the same underlying file as an existing path name
<code>unlink(pathName)</code>	Remove the specified name for a file from its directory; if that was the only name for the underlying file, then remove the file and free its resources.
<code>mkdir(pathName)</code>	Create a new directory with the specified name
<code>rmdir(pathName)</code>	Remove the directory with the specified name
Open and close	
<code>fileDescriptor = open(pathName)</code>	Prepare the calling process for access to the specified file (e.g., check access permissions and initialize kernel data structures for tracking per-process state of open files)
<code>close(fileDescriptor)</code>	Release resources associated with the specified open file
File access	
<code>read(fileDescriptor, buf, len)</code>	Read <code>len</code> bytes from the process's current position in the open file <code>fileDescriptor</code> and copy the results to a buffer <code>buf</code> in the application's memory
<code>write(fileDescriptor, len, buf)</code>	Write <code>len</code> bytes of data from a buffer <code>buf</code> in the process's memory to the process's current position in the open file <code>fileDescriptor</code> .
<code>seek(fileDescriptor, offset)</code>	Change the process's current position in the open file <code>fileDescriptor</code> to the specified <code>offset</code>
<code>dataPtr = mmap(fileDescriptor, off, len)</code>	Set up a mapping between the data in the file <code>fileDescriptor</code> from <code>off</code> to <code>off + len</code> and an area in the application's virtual memory from <code>dataPtr</code> to <code>dataPtr + len</code> .
<code>munmap(dataPtr, len)</code>	Remove the mapping between the application's virtual memory and a mapped file
<code>fsync(fileDescriptor)</code>	Force to disk all buffered, dirty pages for the file associated with <code>fileDescriptor</code> .

Figure 11.6: A simple API for accessing files.

`existingPath` must not be a directory. Why does Linux mandate this restriction?

Answer: Preventing multiple hard links to a directory prevents cycles, ensuring that the directory structure is always a directed acyclic graph (DAG).

Additionally, allowing hard links to a directory would muddle a directory's parent directory entry (e.g., “..” as discussed in the sidebar on page 336.)

Open and close. To start accessing a file, a process calls `open()` to get a *file descriptor* it can use to refer to the open file. *File descriptor* is Unix terminology; in other systems and APIs objects playing similar roles may be called *file handles* or *file streams*.

Definition: **file descriptor**

Definition: **file handle**

Definition: **file stream**

Operating systems require processes to explicitly `open()` files and access them via file descriptors rather than simply passing the path name to `read()` and `write()` calls for two reasons. First, path parsing and permission checking can be done just when a file is opened and need not be repeated on each read or write. Second, when a process opens a file, the operating system creates a data structure that stores information about the process's open file such as the file's ID, whether the process can write or just read the file, and a pointer to the process's current position within the file. The file descriptor can thus be thought of as a reference to the operating system's per-open-file data structure that the operating system will use for managing the process's access to the file.

When an application is done using a file, it calls `close()`, which releases the open file record in the operating system.

File access. While a file is open, an application can access the file's data in two ways. First, it can use the traditional procedural interface, making system calls to `read()` and `write()` on an open file. Calls to `read()` and `write()` start from the process's current file position, and they advance the current file position by the number of bytes successfully read or written. So, a sequence of `read()` or `write()` calls moves sequentially through a file. To support random access within a file, the `seek()` call changes a process's current position for a specified open file.

Rather than using `read()` and `write()` to access a file's data, an application can use `mmap()` to establish a mapping between a region of the process's virtual memory and some region of the file. Once a file has been mapped, memory loads and stores to that virtual memory region will read and write the file's data either by accessing a shared page from the kernel's file cache, or by triggering a page fault exception that causes the kernel to fetch the desired page of data from the

Modern File Access APIs

The API shown in Figure 11.6 is similar to most widely-used file access APIs, but it is somewhat simplified.

For example, each of the listed calls is similar to a call provided by the Posix interface, but the API shown in Figure 11.6 omits some arguments and options found in Posix. The Posix `open()` call, for example, includes two additional arguments one to specify various flags such as whether the file should be opened in read-only or read-write mode and the other to specify the access control permissions that should be used if the `open()` call creates a new file.

In addition, real-world file access APIs are likely to have a number of additional calls. For example, the Microsoft Windows file access API includes dozens of calls including calls to lock and unlock a file, to encrypt and decrypt a file, or to find a file in a directory whose name matches a specific pattern.

file system into memory. When an application is done with a file, it can call `munmap()` to remove the mappings.

Finally, the `fsync()` call is important for reliability. When an application updates a file via a `write()` or a memory store to a mapped file, the updates are buffered in memory and written back to stable storage at some future time. `Fsync()` ensures that all pending updates for a file are written to persistent storage before the call returns. Applications use this function for two purposes. First, calling `fsync()` ensures that updates are durable and will not be lost if there is a crash or power failure. Second, calling `fsync()` between two updates ensures that the first is written to persistent storage before the second. Note that calling `fsync()` is not always necessary; the operating system ensures that all updates are made durable by periodically flushing all dirty file blocks to stable storage.

Exercises

1. **Discussion** Suppose a process successfully opens an existing file that has a single hard link to it, but while the process is reading that file, another process unlinks that file. What should happen to subsequent reads by the first process? Should they succeed? Should they fail? Why?
2. In Linux, suppose a process successfully opens an existing file that has a single hard link to it, but while the process is reading that file, another process unlinks that file? What happens to subsequent reads by the first process? Do they succeed? Do they fail? (Answer this problem by consulting documentation or by writing a program to test the behavior of the system in this case.)

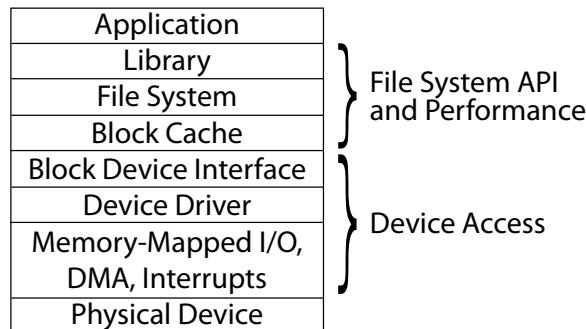


Figure 11.7: Layered abstractions provide access to I/O systems such as storage systems.

3. Write a program that creates a new file, writes 100KB to it, flushes the writes, and deletes it. Time how long each of these steps takes.

Hint You may find the Posix system calls `creat()`, `write()`, `fflush()`, `close()`, and `gettimeofday()` useful. See the manual pages for details on how to use these.

4. Consider a text editor that saves a file whenever you click a save button. Suppose that when you press the button, the editor simply (1) animates the button “down” event (e.g., by coloring the button grey), (2) uses the `write()` system call to write your text to your file, and then (3) animates the button “up” event (e.g., by coloring the button white). What bad thing could happen if a user edits a file, saves it, and then turns off her machine by flipping the power switch (rather than shutting the machine down cleanly)?

11.3 Software layers

As shown in Figure 11.7, operating systems implement the file system abstraction through a series of software layers. Broadly speaking, these layers have two sets of tasks:

- **API and performance.** The top levels of the software stack—user-level libraries, kernel-level file systems, and the kernel’s block cache—provide a convenient API for accessing named files and also work to minimize slow storage accesses via caching, write buffering, and prefetching.

- **Device access.** Lower levels of the software stack provide ways for the operating system to access a wide range of I/O devices. Device drivers hide the details of specific I/O hardware by providing hardware-specific code for each device, and placing that code behind a simpler, more general interfaces that the rest of the operating system can use such as a block device interface. The device drivers execute as normal kernel-level code, using the systems' main processors and memory, but they must interact with the I/O devices. A system's processors and memory communicate with its I/O devices using Memory-Mapped I/O, DMA, and Interrupts.

In the rest of this section, we first talk about the file system API and performance layers. We then discuss device access.

11.3.1 API and performance

The top levels of the file system software stack—divided between application libraries and operating system kernel code—provide the file system API and also provide caching and write buffering to improve performance.

System calls and libraries. The file system abstraction such as the API shown in Figure 11.6 can be provided by directly by system calls. Alternatively, application libraries can wrap the system calls to add additional functionality such as buffering.

For example, in Linux, applications can access files directly using system calls (e.g., `open()`, `read()`, `write()`, and `close()`.) Alternatively, applications can use the `stdio` library calls (e.g., `fopen()`, `fread()`, `fwrite()`, and `fclose()`). The advantage of the latter is that the library includes buffers to aggregate a program's small reads and writes into system calls that access larger blocks, which can reduce overheads. For example, if a program uses the library function `fread()` to read 1 byte of data, the `fread()` implementation may use the `read()` system call to read a larger block of data (e.g., 4 KB) into a buffer maintained by the library in the application's address space. Then, if the process calls `fread()` again to read another byte, the library just returns the byte from the buffer without needing to do a system call.

Block cache. Typical storage devices are much slower than a computer's main memory. The operating system's block cache therefore caches recently read blocks, and it buffers recently written blocks so that they can be written back to the storage device at a later time.

In addition to improving performance by caching and write buffering, the block cache serves as a synchronization point: because all requests for a given block go through the block cache, the operating system includes information

with each buffer cache entry to, for example, prevent one process from reading a block while another process writes it or to ensure that a given block is only fetched from the storage device once, even if it is simultaneously read by many processes.

Prefetching. Operating systems use prefetching to improve I/O performance. For example, if a process reads the first two blocks of a file, the operating system may prefetch the next ten blocks.

Such prefetching can have several beneficial effects:

- **Reduced latency.** When predictions are accurate, prefetching can help the latency of future requests because reads can be serviced from main memory rather than from slower storage devices.
- **Reduced device overhead.** Prefetching can help reduce storage device overheads by replacing a large number of small requests with one large one.
- **Improved parallelism.** Some storage devices such as Redundant Arrays of Inexpensive Disks (RAIDs) and Flash drives are able to process multiple requests at once, in parallel. Prefetching provides a way for operating systems to take advantage of available hardware parallelism.

Prefetching, however, must be used with care. Too-aggressive prefetching can cause problems:

- **Cache pressure.** Each prefetched block is stored in the block cache, and it may displace another block from the cache. If the evicted block is needed before the prefetched one is used, prefetching is likely to hurt overall performance.
- **I/O contention.** Prefetch requests consume I/O resources. If other requests have to wait behind prefetch requests, prefetching may hurt overall performance.
- **Wasted effort.** Operating system prefetching is speculative. If the prefetched blocks end up being needed, then prefetching can help performance; otherwise, prefetching may hurt overall performance by wasting memory space, I/O device bandwidth, and CPU cycles.

11.3.2 Device drivers: Common abstractions

Device drivers translate between the high level abstractions implemented by the operating system and the hardware-specific details of I/O devices. Definition: **device driver**

Challenge: Device Driver Reliability

Because device drivers are hardware-specific, they are often written and updated by the hardware manufacturer rather than the operating system's main authors. Furthermore, because there are large numbers of devices—some operating systems support tens of thousands of devices—device driver code may represent a large fraction of an operating system's code.

Unfortunately, bugs in device drivers have the potential to affect more than the device. A device driver usually runs as part of the operating system kernel since kernel routines depend on it and because it needs to access the hardware of its device. However, if the device driver is part of the kernel, then a device driver's bugs have the potential to affect the overall reliability of a system. For example, in 2003 it was reported that drivers caused about 85% of failures in the Windows XP operating system.

To improve reliability, operating systems are increasingly using protection techniques similar to those used to isolate user-level programs to isolate device drivers from the kernel and from each other.

An operating system may have to deal with many different I/O devices. For example, a laptop on a desk might be connected to two keyboards (one internal and one external), a trackpad, a mouse, a wired ethernet, a wireless 802.11 network, a wireless bluetooth network, two disk drives (one internal and one external), a microphone, a speaker, a camera, a printer, a scanner, and a USB thumb drive. And that is just a handful of the literally thousands of devices that could be attached to a computer today. Building an operating system that treats each case separately would be impossibly complex.

Layering helps simplify operating systems by providing common ways to access various classes of devices. For example, for any given operating system, storage device drivers typically implement a standard *block device* interface that allows data to be read or written in fixed-sized blocks (e.g., 512, 2048, or 4096 bytes).

Definition: **block device**

Such a standard interface lets an operating system easily use a wide range of similar devices. A file system implemented to run on top of the standard block device interface can store files on any storage device whose driver implements that interface, be it a Seagate spinning disk drive, an Intel solid state drive, a Western Digital RAID, or an Amazon Simple Block Store volume. These devices all have different internal organizations and control registers, but if each manufacturer provides a device driver that exports the standard interface, the rest of the operating system does not need to be concerned with these per-device details.

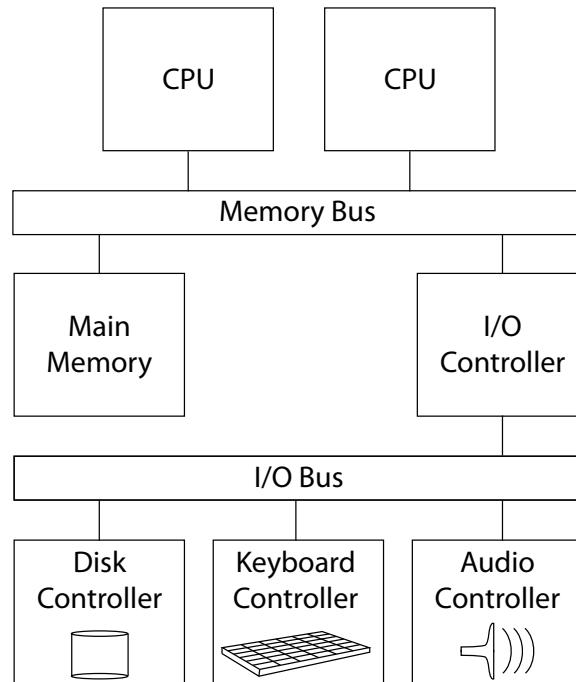


Figure 11.8: I/O devices are attached to the I/O bus, which is attached to the memory bus.

11.3.3 Device access

How should an operating system's device drivers communicate with and control a storage device? At first blush, a storage device seems very different from the memory and CPU resources we have discussed so far. For example, a disk drive includes several motors, a sensor for reading data, and an electromagnet for writing data.

Memory mapped I/O. As Figure 11.8 illustrates, I/O devices are typically connected to an I/O bus that is connected to the system's memory bus. Each I/O device has controller with a set of registers that can be written and read to transmit commands and data to and from the device. For example, a simple keyboard controller might have one register that can be read to learn the most recent key pressed and another register than can be written to turn the caps-lock light on or off.

To allow I/O control registers to be read and written, systems implement memory mapped I/O. *Memory mapped I/O* maps each device's control registers to a range of physical addresses on the memory bus. Reads and writes by the CPU to this physical address range do not go to main memory. Instead, they

Definition: **Memory mapped I/O**

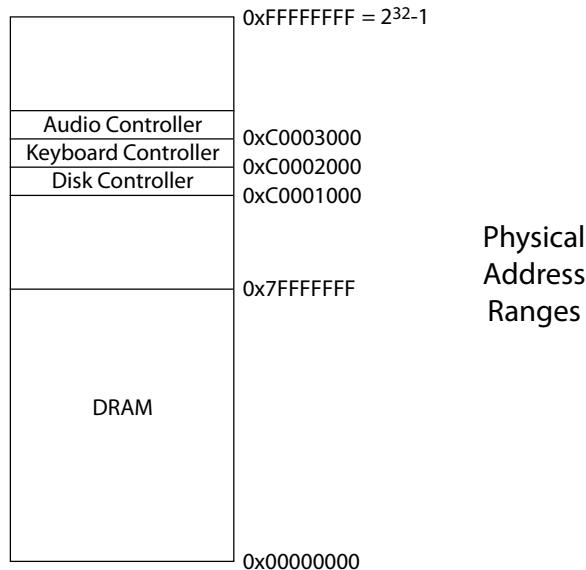


Figure 11.9: Physical address map for a system with 2 GB of DRAM and 3 memory mapped I/O devices.

go to registers on the I/O devices's controllers. Thus, the operating system's keyboard device driver might learn the value of the last key pressed by reading from physical address, say, $0xC00002000$.

The hardware maps different devices to different physical address ranges. Figure 11.9 shows the physical address map for a hypothetical system with a 32 bit physical address space capable of addressing 4 GB of physical memory. This system has 2 GB of DRAM in it, consuming physical addresses $0x00000000$ (0) to $0x7FFFFFFF$ ($2^{31} - 1$). Controllers for each of its three I/O devices are mapped to ranges of addresses in the first few kilobytes above 3 GB. For example, physical addresses from $0xC0001000$ to $0xC0001FFF$ access registers in the disk controller.

DMA. Many I/O devices, including most storage devices, transfer data in bulk. For example, operating systems don't read a word or two from disk, they usually do transfers of at least a few kilobytes at a time. Rather than requiring the CPU to read or write each word of a large transfer, I/O devices can use direct memory access. When using *direct memory access (DMA)*, the I/O device copies a block of data between its own internal memory and the system's main memory.

Definition: **direct memory access (DMA)**

To set up a DMA transfer, a simple operating system could use memory mapped I/O to provide a target physical address, transfer length, and operation code to the device. Then, the device copies data to or from the target address

Port mapped I/O

Today, memory mapped I/O is the dominant paradigm for accessing I/O device's control registers. However an older style, *port mapped I/O*, is still sometimes used. Notably, the x86 architecture supports both memory mapped I/O and port mapped I/O.

Port mapped I/O is similar to memory mapped I/O in that instructions read from and write to specified addresses to control I/O devices. There are two differences. First, where memory mapped I/O uses standard memory-access instructions (e.g., `load` and `store`) to communicate with devices, port mapped I/O uses distinct I/O instructions. For example, the x86 architecture uses the `in` and `out` instructions for port mapped I/O. Second, whereas memory mapped I/O uses the same physical address space as is used for the system's main memory, the address space for port mapped I/O is distinct from the main memory address space.

For example, in x86 I/O can be done using either memory mapped or port mapped I/O, and the low-level assembly code is similar for both cases:

Memory mapped I/O

```
MOV register, memAddr // To read
MOV memAddr, register // To write
```

Port mapped I/O

```
IN register, portAddr // To read
OUT portAddr, register // To write
```

Port mapped I/O can be useful in architectures with constrained physical memory addresses since I/O devices do not need to consume ranges of physical memory addresses. On the other hand, for systems with sufficiently large physical address spaces, memory mapped I/O can be simpler since no new instructions or address ranges need to be defined and since device drivers can use any standard memory access instructions to access devices. Also, memory mapped I/O provides a more unified model for supporting DMA—direct transfers between I/O devices and main memory.

without requiring additional processor involvement.

After setting up a DMA transfer, the operating system must not use the target physical pages for any other purpose until the DMA transfer is done. The operating system therefore “pins” the target pages in memory so that they can not be reused until they are unpinned. For example, a pinned physical page cannot be swapped out to disk and then remapped to some other virtual address.

Interrupts. The operating system needs to know when I/O devices have completed handling a request or when new external input arrives. One option is *polling*, repeatedly using memory mapped I/O to read a status register on the device. Because I/O devices are often much slower than CPUs and because inputs received by I/O devices may arrive at irregular rates, it is usually better for I/O devices to use interrupts to notify the operating system of important

Definition: **polling**

Advanced DMA

Although setting up a device's DMA can be as simple as providing a target physical address and length and then saying "go!", more sophisticated interfaces are increasingly used.

For example rather than giving devices direct access to the machine's physical address space, some systems include an *I/O memory management unit (IOMMU)* that translates device virtual addresses to main memory physical addresses similar to how a processor's TLB translates processor virtual addresses to main memory physical addresses. An IOMMU can provide both protection (e.g., preventing a buggy IO device from overwriting arbitrary memory) and simpler abstractions (e.g., allowing devices to use virtual addresses so that, for example, a long transfer can be made to a range of consecutive virtual pages rather than a collection of physical pages scattered across memory.)

Also, some devices add a level of indirection so that they can interrupt the CPU less often. For example, rather than using memory mapped I/O to set up each DMA request, the CPU and I/O device could share two lists in memory: one list of pending I/O requests and another of completed I/O requests. Then, the CPU could set up dozens of disk requests and only be interrupted when all of them are done.

Sophisticated I/O devices can even be configured to take different actions depending on the data they receive. For example, some high performance network interfaces parse incoming packets and direct interrupts to different processors based on the network connection to which a received packet belongs.

events.

11.3.4 Putting it all together: A simple disk request

When a process issues a system call like `read()` to read data from disk into the process's memory, the operating system moves the calling thread to a wait queue. Then, the operating system uses memory mapped I/O both to tell the disk to read the requested data and to set up DMA so that the disk can place that data in the kernel's memory. The disk then reads the data and DMAs it into main memory; once that is done, the disk triggers an interrupt. The operating system's interrupt handler then copies the data from the kernel's buffer into the process's address space. Finally, the operating system moves the thread to the ready list. When the thread next runs, it will return from the system call with the data now present in the specified buffer.

Exercises

5. Write a program that times how long it takes to issue 100,000 one-byte

writes in each of two ways. First, time how long it takes to use the Posix system calls `creat()`, `write()`, and `close()` directly. Then see how long these writes take if the program uses the stdio library calls (e.g., `fopen()`, `fwrite()`, and `fclose()`) instead. Explain your results.

11.4 Conclusions and future directions

The file system interface is a stable one, and small variations of interface described here can be found in many operating systems and for many storage devices.

Yet, the file system abstraction is imperfect, and application writers need to use it carefully to get acceptable performance and reliability. For example, if an application `write()`s a file, the update may not be durable when the `write()` call returns; application writers often call `fsync()` to ensure durability of data.

Could better file system APIs simplify programming? For example, if file systems allowed users to update multiple objects atomically, that might simplify many applications that currently must carefully constrain the order that their updates are stored using crude techniques such as using `fsync` as a barrier between one set of updates and the next.

Could better file system APIs improve performance? For example, one proposed interface allows an application to direct the operating system to transfer a range of bytes from a file to a network connection. Such an interface might, for example, reduce overheads for a movie server that streams movies across a network to clients.

Exercises

For convenience, the exercises from the body of the chapter are repeated here.

1. **Discussion** Suppose a process successfully opens an existing file that has a single hard link to it, but while the process is reading that file, another process unlinks that file. What should happen to subsequent reads by the first process? Should they succeed? Should they fail? Why?
2. In Linux, suppose a process successfully opens an existing file that has a single hard link to it, but while the process is reading that file, another process unlinks that file? What happens to subsequent reads by the first process? Do they succeed? Do they fail? (Answer this problem by consulting documentation or by writing a program to test the behavior of the system in this case.)
3. Write a program that creates a new file, writes 100KB to it, flushes the writes, and deletes it. Time how long each of these steps takes.
Hint You may find the Posix system calls `creat()`, `write()`, `fflush()`, `close()`, and `gettimeofday()` useful. See the manual pages for details on how to use these.
4. Consider a text editor that saves a file whenever you click a save button. Suppose that when you press the button, the editor simply (1) animates the button “down” event (e.g., by coloring the button grey), (2) uses the `write()` system call to write your text to your file, and then (3) animates the button “up” event (e.g., by coloring the button white). What bad thing could happen if a user edits a file, saves it, and then turns off her machine by flipping the power switch (rather than shutting the machine down cleanly)?
5. Write a program that times how long it takes to issue 100,000 one-byte writes in each of two ways. First, time how long it takes to use the Posix system calls `creat()`, `write()`, and `close()` directly. Then see how long these writes take if the program uses the stdio library calls (e.g., `fopen()`, `fwrite()`, and `fclose()`) instead. Explain your results.

Chapter 12

Storage Devices

Treat disks like tape

– John Ousterhout

Although today’s persistent storage devices have large capacity and low cost, they have drastically worse performance than volatile DRAM memory.

Not only that, but the characteristics are different and are peculiar to specific persistent storage devices. For example, although programs can access random individual words of DRAM with good performance, programs can only access today’s disk and flash storage devices hundreds or thousands of bytes at a time. Furthermore, even if an application restricts itself to supported access sizes (e.g., 2 KB per read or write), if the application accesses pattern is random, the application may be slower by a factor of several hundred than if the application accessed the same amount of data sequentially.

To cope with the limitations and to maximize the performance of storage devices, both file system designers and application writers need to understand the physical characteristics of persistent storage devices.

Roadmap. This chapter discusses two types of persistent storage, *magnetic disks* and *flash memory*. Both are widely used: magnetic disks provide persistent storage for most servers, workstations, and laptops, while flash memory provides persistent storage for most smart phones, tablets, and cameras and for an increasing fraction of laptops.



Figure 12.1: A partially-disassembled magnetic disk drive.

12.1 Magnetic disk

Magnetic disk is a nonvolatile storage technology that is widely used in laptops, desktops, and servers. Disk drives work by magnetically storing data on a thin metallic film bonded to a glass, ceramic, or aluminum disk that rotates rapidly. Figure 12.1 shows a disk drive without its protective cover, and Figure 12.2 shows a schematic of a disk drive, identifying key components.

Definition: **platters** Each drive holds one or more *platters*, thin cylinders that hold the magnetic material. Each platter has two *surfaces*, one on each side. When the drive is powered up, the platters are constantly spinning on a *spindle* powered by a *motor*. In 2011, disks commonly spin at 4200–15000 RPM (70–250 revolutions per second.)

Definition: **head** A disk *head* is the component that reads and writes data by sensing or introducing a magnetic field on a surface. There is one head per surface, and as a surface spins underneath a head, the head reads or writes a sequence of bits along a circle centered on the disk's spindle. As a disk platters spins, it creates a layer of rapidly spinning air, and the disk head floats on that layer, allowing the head to get extremely close to the platter without contacting it. A *head crash* occurs when the disk head breaks through this layer with enough force to damage the magnetic surface below; head crashes can be caused by excessive shock such as dropping a running drive.

Definition: **arm** In order to use the full surface, each head is attached to an *arm*, and all of Definition: **arm assembly** a disk's arms are attached to a single *arm assembly* that includes a motor that

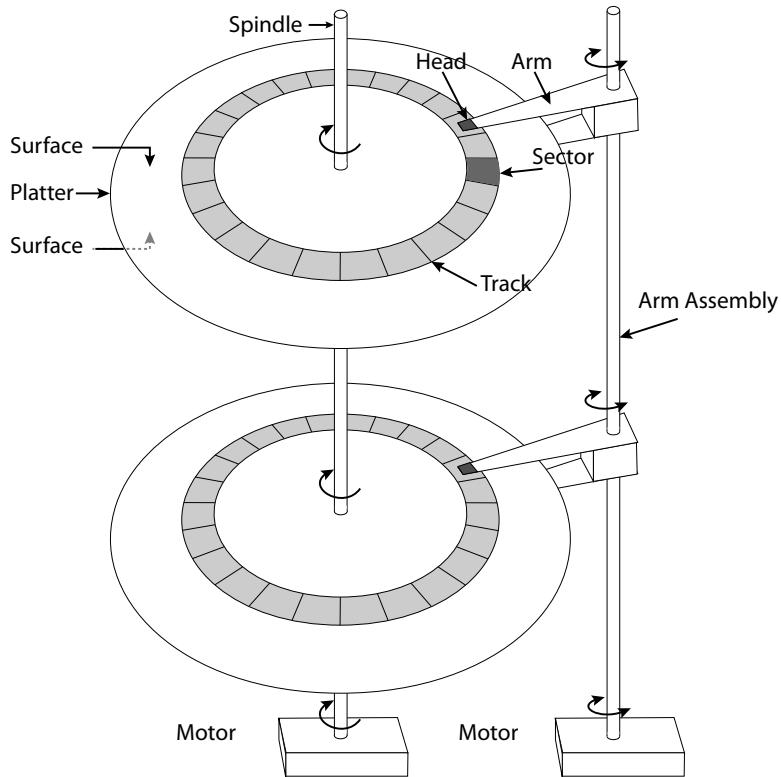


Figure 12.2: Key components of a magnetic disk drive.

can move the arms across the surfaces of the platters. Note that an assembly has just one motor, and all of its arms move together.

Data bits are stored in fixed-size *sectors*; typically sectors are 512 bytes. The disk hardware cannot read or write individual bytes or words; instead, it must always read or write at least an entire sector. This means that to change 1 byte in a sector, the operating system must read the old sector, update the byte in memory, and rewrite the entire sector to disk. One reason for this restriction is that the disk encodes each sector with additional error correction code data, allowing it to fix (or at least detect) imperfectly read or written data, which, in turn allows higher density storage and higher bandwidth operation.

A circle of sectors on a surface is called a *track*. All of the data on a track can be read or written without having to move the disk arm, so reading or writing a sequence of sectors on the same track is much faster than reading or writing sectors on different tracks.

To maximize sequential access speed, logical sector zero on each track is staggered from sector zero on the previous track by an amount corresponding

to time it takes the disk to move the head from one track to another or to switch from the head on one surface to the head on another one. This staggering is called *track skewing*.

Definition: **track skewing**

To increase storage density and disk capacity, disk manufacturers make tracks and sectors as thin and small as possible. If there are imperfections in a sector, then that sector may be unable to reliably store data. Manufacturers therefore include spare sectors distributed across each surface. The disk

Definition: **sector sparing** firmware or the file system's low-level formatting can then use *sector sparing* to

Definition: **slip sparing** remap sectors to use spare sectors instead of faulty sectors. *Slip sparing* helps retain good sequential access performance by remapping all sectors from the bad sector to the next spare, advancing each logical sector in that range by one physical sector on disk.

Definition: **buffer memory**

Disk drives often include a few MB of *buffer memory*, memory that the disk's controller uses to buffer data being read from or written to the disk, for track buffering, and for write acceleration.

Definition: **track buffer**

Track buffering improves performance by storing sectors that have been read by the disk head but not yet requested by the operating system. In particular, when a disk head moves to a track, it may have to wait for the sector it needs to access to rotate under the disk head. While the disk is waiting, it reads unrequested sectors to its track buffer so that if the operating system requests those sectors later, they can be returned immediately.

Definition: **write acceleration**

Write acceleration stores data to be written to disk in the disk's buffer memory and acknowledges the writes to the operating system before the data is actually written to the platter; the disk firmware flushes the writes from the track buffer to the platter at some later time. This technique can significantly increase the apparent speed of the disk, but it carries risks—if power is lost before buffered data is safely stored, then data might be lost.

Server drives implementing the SCSI or Fibre Channel interfaces and increasing numbers of commodity drives with the Serial ATA (SATA) interface

Definition: **tag command queuing**

implement a safer form of write acceleration with *tag command queuing* (TCQ)

Definition: **native command queuing**

(also called *native command queuing* (NCQ) for SATA drives.) TCQ allows an operating system to issue multiple concurrent requests to the disk and for the disk to process those requests out of order to optimize scheduling, and it can be configured to only acknowledge write requests when the blocks are safely on the platter.

12.1.1 Disk access and performance

Operating systems send commands to a disk to read or write one or more consecutive sectors. A disk's sectors are identified with *logical block addresses* (LBAs)

Definition: **logical block addresses**

that specify the surface, track, and sector to be accessed.

To service a request for a sequence of blocks starting at some sector, the disk

must first seek to the right track, then wait for the first desired sector to rotate to the head, and then transfer the blocks. So, the time for a disk access is:

$$\text{disk access time} = \text{seek time} + \text{rotation time} + \text{transfer time}$$

- **Seek.** The disk must first *seek*—move its arm over the desired track. Definition: **seek**
To seek, the disk first activates a motor that moves the arm assembly to approximately the right place on disk. Then, as arm stops vibrating from the motion of the seek, the disk begins reading positioning information embedded in the sectors to determine exactly where it is and to make fine-grained positioning corrections to *settle* on the desired track. Once Definition: **settle** the head has settled on the right track, the disk uses signal strength and positioning information to make minute corrections in the arm position to keep the head over the desired track.

A request's seek time depends on how far the disk arm has to move.

A disk's *minimum seek time* is the time it takes for the head to move Definition: **minimum seek time** from one track to an adjacent one. For short seeks, disks typically just “reset” the head on the new track by updating the target track number in the track-following circuitry. Minimum seek times of 0.3–1.5 ms are typical.

If a disk is reading the t th track on one surface, its *head switch time* is the time it would take to begin reading the t th track on a different surface. Definition: **head switch time** Tracks can be less than a micron wide and tracks on different surfaces are not perfectly aligned. So, a head switch between the same tracks on different surfaces has a cost similar to a minimum seek: the disk begins using the sensor on a different head and then resettles the disk on the desired track for that surface.

A disk's *maximum seek time* is the time it takes the head to move from Definition: **maximum seek time** the innermost track to the outermost one or vice versa. Maximum seek times are typically over 10 ms and can be over 20 ms.

A disk's *average seek time* is the average across seeks between each possible pair of tracks on a disk. This value is often approximated as the time to seek one third of the way across the disk. Definition: **average seek time**

- **Rotate.** Once the disk head has settled on the right track, it must wait for the target sector to rotate under it. This waiting time is called the *rotational latency*. Today, most disks rotate at 4200 RPM to 15,000 RPM (15 ms to 4 ms per rotation), and for many workloads a reasonable estimate of rotational latency is one-half the time for a full rotation—7.5 ms–2 ms. Definition: **rotational latency**

Once a disk head has settled on a new track, most disks immediately begin reading sectors into their buffer memory, regardless of which sectors have been requested. This way, if there is a request for one of the sectors that

Beware of “average seek time”

Although the name *average seek time* makes it tempting to use this metric when estimating the time it will take a disk to complete a particular workload, it is often the wrong metric to use. Average seek time—the average across seeks between each possible pair of tracks on disk—was defined this way to make it a well-defined, standard metric, not because it is representative of common workloads.

The definition of average seek time essentially assumes no locality in a workload, so it is very nearly a worst-case scenario. Many workloads access sectors that are likely to be near one another; for example, most operating systems attempt to place files sequentially on disk and to place different files in a directory on the same track or on tracks near one another. For these (common) workloads, the seek times observed may be closer to the disk’s minimum seek time than its “average” seek time.

The demise of the cylinder

A *cylinder* on a disk is a set of tracks on different surfaces with the same track index. For example, on a 2-platter drive, the 8th tracks on surfaces 0, 1, 2, and 3 would form the 8th cylinder of the drive.

Some early file systems put related data on different surfaces but in the same cylinder. The idea was that data from the different tracks in the cylinder could be read without a requiring a seek. Once a cylinder was full, the file system would start placing data in one of the adjacent cylinders.

As disk densities have increased, the importance of the cylinder has declined. Today, a disk’s tracks can be less than a micron wide. To follow a track at these densities, a controller monitors the signals from a disk’s head to control the disk arm assembly’s motor to keep the head centered on a track. Furthermore, at these densities, the tracks of a cylinder may not be perfectly aligned. As a result, when a disk switches disk heads, the new head must center itself over the desired track. So, switching heads within a cylinder ends up being similar to a short 1-track seek: the controller chooses the new cylinder/track and the disk head settles over the target track. Today, accessing different tracks within the same cylinder costs about the same as accessing adjoining tracks on the same platter.

have already passed under the disk head, the data can be transferred immediately, rather than having to delay the request for nearly a full rotation to reread the data.

- **Transfer.** Once the disk head reaches a desired sector, the disk must transfer the data from the sector to its buffer memory (for reads) or vice versa (for writes) as the sectors rotate underneath the head. Then, for reads, it must transfer the data from its buffer memory to the host's main memory. For writes, the order of the transfers is reversed.

To amortize seek and rotation time, disk requests are often for multiple sequential sectors. The time to transfer one or more sequential sectors from (or to) a surface once the disk head begins reading (or writing) the first sector is the *surface transfer time*.

On a modern disk, the surface transfer time for a single sector is much smaller than the seek time or rotational latency. For example, disk bandwidths often exceed 100 MB/s, so the surface transfer time for a 512-byte sector is often under 5 microseconds (0.005 ms).

Because a disk's outer tracks have room for more sectors than its inner tracks and because a given disk spins at a constant rate, the surface transfer bandwidth is often higher for the outer tracks than the inner tracks.

For a disk read, once sectors have been transferred to the disk's buffer memory, they must be transferred to the host's memory over some connection such as SATA (serial ATA), SAS (serial attached SCSI), Fibre Channel, or USB (universal serial bus). For writes, the transfer goes in the other direction. The time to transfer data between the host's memory and the disk's buffer is the *host transfer time*. Typical bandwidths range from 60MB/s for USB 2.0 to 2500 MB/s for Fibre Channel-20GFC.

For multi-sector reads, disks pipeline transfers between the surface and disk buffer memory and between buffer memory and host memory; so for large transfers, the total transfer time will be dominated by whichever of these is the bottleneck. Similarly, for writes, disks overlap the host transfer with the seek, rotation, and surface transfer; again, the total transfer time will be dominated by whichever is the bottleneck.

Definition: **surface transfer time**

Definition: **host transfer time**

Example: Toshiba MK3254GSY. Figure 12.3 shows some key parameters for a recent 2.5-inch disk drive for laptop computers.

This disk stores 320 GB of data on 2 platters, so it stores 80 GB per surface. The platters spin at 7200 revolutions per minute, which is 8.3 ms per revolution; since each platter's diameter is about 6.3 cm, the outer edge of each platter is moving at about 85 km/hour!

The disk's data sheet indicates an average seek time for the drive of 10.5 ms for reads and 12.0 ms for writes. The seek time for reads and writes differs be-

Size	
Platters/Heads	2/4
Capacity	320 GB
Performance	
Spindle speed	7200 RPM
Average seek time read/write	10.5 ms / 12.0 ms
Maximum seek time	19 ms
Track-to-track seek time	1 ms
Transfer rate (surface to buffer)	54–128 MB/s
Transfer rate (buffer to host)	375 MB/s
Buffer memory	16 MB
Power	
Typical	16.35 W
Idle	11.68 W

Figure 12.3: Hardware specifications for a laptop disk (Toshiba MK3254GSY) manufactured in 2008.

cause the disk starts attempting to read data before the disk arm has completely settled, but it must wait a bit longer before it is safe to write.

When transferring long runs of contiguous sectors, the disk's bandwidth is 54–128 MB/s. The bandwidth is expressed as a range because the disk's outer tracks have more sectors than its inner tracks, so when the disk is accessing data on its outer tracks, sectors sweep past the disk head at a higher rate.

Once the data is transferred off the platter, the disk can send it to the main memory of the computer at up to 375 MB/s via a SATA (Serial ATA) interface.

Random v. sequential performance. Given seek and rotational times measured in milliseconds, small accesses to random sectors on disk are much slower than large, sequential accesses.

Example: Random access workload.

Question: For the disk described in Figure 12.3, consider a workload consisting of 500 read requests, each of a randomly chosen sector on disk, assuming requests are serviced in FIFO order. How long will servicing these requests take?

Answer: Disk access time is seek time + rotation time + transfer time.

Seek time. Each request requires a seek from a random starting track to a random ending track, so the disk's average seek time of 10.5 ms is a good estimate of the cost of each seek.

Rotation time. Once the disk head settles on the right track, it must wait for the desired sector to rotate under it; since there is no reason to expect the desired sector to be particularly near or far from the disk head when it settles, a reasonable estimate for rotation time is 4.15 ms, one half of the time that it takes a 7200 RPM disk to rotate once.

Transfer time. The disk's surface bandwidth is at least 54 MB/s, so transferring 512 bytes takes at most $9.5 \mu\text{s}$ (0.0095 ms).

Total time. $10.5 + 4.15 + .0095 = 14.65$ ms per request, so 500 requests will take about 7.8 seconds.

Example: Sequential access workload.

Question: For the disk described in Figure 12.3, consider a workload consisting of a read request for 500 sequential sectors on the same track. How long will servicing these requests take?

Answer: Disk access time is seek time + rotation time + transfer time.

Seek time. Since we don't know which track we're starting with or which track we're reading from, we'll use the average seek time, 10.5 ms, as an estimate for the seek time.

Rotation time. Since we don't know the position of the disk when the request is issued, a simple and reasonable estimate for the time for the first desired block to rotate to the disk head is 4.15 ms, one half of the time that it takes a 7200 RPM disk to rotate once.

Transfer time. A simple answer is that 500 sectors can be transferred in 4.8 to 2.0 ms, depending on whether they are on the inner or outer tracks.

$$500 \text{ sectors} * 512 \frac{\text{bytes}}{\text{sector}} * \frac{\text{second}}{54 * 10^6 \text{ bytes}} = 4.8 \text{ ms}$$

$$500 \text{ sectors} * 512 \frac{\text{bytes}}{\text{sector}} * \frac{\text{second}}{128 * 10^6 \text{ bytes}} = 2 \text{ ms}$$

(Too) simple answer. These three estimates give us a range from

$$10.5 + 4.15 + 2 = 16.7 \text{ ms}$$

to

$$10.5 + 4.15 + 4.8 = 19.5 \text{ ms}$$

More precise answer. However, this simple answer ignores the track buffer. Since the transfer time is a large fraction of the rotation time (about 1/4 to 1/2 of the time for a full rotation), we

know that the request covers a significant fraction of a track. This means that there is a good chance that after the seek and settle time, the disk head will be in the middle of the region to be read. In this case, the disk will immediately read some of the track into the track buffer; then it will wait for the first track to rotate around; then it will read the remainder of the desired data.

We can estimate that for the outer track, there is a one in four chance that the initial seek and settle will finish while the head is within the desired range of sectors, and that when that happens, we read an average of $\frac{1}{8}$ th of the desired data before we arrive at the first desired sector. So, for the outer track, this overlap will save us $\frac{1}{4} * \frac{1}{8} = \frac{1}{32}$ of a rotation for the average transfer. This effect slightly reduces the average access time: $16.7 \text{ ms} - \frac{1}{32} * 8.3 \text{ ms} = 16.4 \text{ ms}$.

Similarly, for the inner tracks, there is about a one in two chance that the initial seek will settle in the middle of the desired data, saving on average $\frac{1}{2} * \frac{1}{4} = \frac{1}{8}$. This reduces the average access time: $19.5 \text{ ms} - \frac{1}{8} * 8.3 \text{ ms} = 18.5 \text{ ms}$.

So, we estimate that such an access would take between 16.4 ms and 18.5 ms.

Notice that the sequential workload takes vastly less time than the random workload (less than 20 milliseconds v. 5.5 seconds). This orders of magnitude disparity between sequential and random access performance influences many aspects of file system design and use.

Still, even for the 500 sector request, a non-trivial amount of the access time is spent seeking and rotating rather than transferring.

Example: Effective bandwidth.

Question: For the transfer of 500 sequential sectors examined in the previous example, what fraction of the disk's surface bandwidth is realized?

Answer: The effective bandwidth ranges from

$$(500 \text{ sectors})(512 \frac{\text{bytes}}{\text{sector}})(\frac{1}{18.5 \text{ ms}})(\frac{1 \text{ MB}}{1000000 \text{ bytes}})(\frac{1000 \text{ ms}}{\text{s}}) = 13.8 \text{ MB/s}$$

$$(500 \text{ sectors})(512 \frac{\text{bytes}}{\text{sector}})(\frac{1}{16.4 \text{ ms}})(\frac{1 \text{ MB}}{1000000 \text{ bytes}})(\frac{1000 \text{ ms}}{\text{s}}) = 15.6 \text{ MB/s}$$

This gives us a range of $\frac{13.8 \text{ MB/s}}{54 \text{ MB/s}} = 26\%$ to $\frac{15.6 \text{ MB/s}}{128 \text{ MB/s}} = 12\%$ of the maximum bandwidth from the inner to the outer tracks.

So, even a fairly large request (500 sectors or 250KB in this case) can incur significant overheads from seek and rotational latency.

Example: Efficient access.

Question: For the disk described in Figure 12.3, how large must a request that begins on a random disk sector be to ensure that the disk gets at least 80% of its advertised maximum surface transfer bandwidth?

Answer: When reading a long sequence of logically sequential blocks, the disk will read an entire track, then do a 1 track seek (or a head switch and reset, which amounts to the same thing) and then read the next track. Notice that track buffering allows the disk to read an entire track in one rotation regardless of which sector the head is over when it settles on the track and starts successfully reading. So, for the outer tracks, it reads for one rotation (8.4 ms) and then does a minimum seek (1 ms).

So, to achieve 80% of peak bandwidth after a random seek (10.5 ms), we need to read enough rotations worth of data to ensure that we spend 80% of the total time reading. If x is the number of rotations we will read, then we have

$$\begin{aligned} 0.8 \text{ totalTime} &= x \text{ rotationTime} \\ 0.8(10.5 \text{ ms} + (1 + 8.4)x\text{ms}) &= 8.4x \text{ ms} \\ x &= 9.09 \end{aligned}$$

So, we need to read at least 9.09 rotations worth of data to reach an efficiency of 80%. Since each rotation takes 8.4 ms and transfers data at 128 MB/s, 9.09 rotations transfers 9.77 MB of data, or about 19,089 sectors.

12.1.2 Disk scheduling

Because moving the disk arm and waiting for the platter to rotate are so expensive, performance can be significantly improved by optimizing the order in which pending requests are serviced. Disk scheduling can be done by the operating system, by the disk's firmware, or both.

FIFO. The simplest thing to do is to process requests in first-in-first-out (FIFO) order. Unfortunately, a FIFO scheduler can yield poor performance. For example, a sequence of requests that alternate between the outer and inner tracks of a disk will result in many long seeks.

SPTF/SSTF. An initially appealing option is to use a greedy scheduler that, given the current position of the disk head and platter, always services the pending request that can be serviced in the minimum amount of time. This approach is called *shortest positioning time first* (SPTF) (or *shortest seek time first* (SSTF) if rotational positioning is not considered.)

Definition: **shortest positioning time first**
Definition: **shortest seek time first**

SPTF and SSTF have two significant limitations. First, because moving the disk arm and waiting for some rotation time affects the cost of serving subsequent requests, these greedy approaches are not guaranteed to optimize disk performance. Second, these greedy approaches can cause starvation when, for example, a continuous stream of requests to inner tracks prevents requests to outer tracks from ever being serviced.

Example: SPTF is not optimal.

Question: Suppose a disk's head is just inside the middle track of a disk so that seeking to the inside track would cost 9.9 ms while seeking to the outside track would cost 10.1 ms. Assume that for the disk in question, seeking between the outer and inner track costs 15 ms and that a rotation takes 10ms.

Also suppose that the disk has two sets of pending requests. The first set is 1000 requests to read each of the 1000 sectors on the inner track of the disk; the second set is 2000 requests to read each of the 2000 sectors on the outer track of the disk.

Compare the average response time per request for the SPTF schedule (first read the "nearby" inner track and then read the outer track) and the alternative of reading the outer track first and then the inner track.

Answer: To service either the outer set of requests, the disk must seek to the appropriate track and then wait for one full rotation while all of the track's data sweeps under the arm. For either set, the average response time for a request in that set will be the delay until the seek completes plus one half the disk's rotation time. Notice that the set is handled second must wait until the first one is completely done before it can start, adding to the response time observed for requests in that set.

$$\begin{aligned} \text{Inner first (SPTF): } & (1000(9.9 + 5) + 2000(9.9 + 10 + 15 + 5))/3000 = 31.6 \text{ ms} \\ \text{Outer first: } & (2000(10.1 + 5) + 1000(10.1 + 10 + 15 + 5))/3000 = 23.3 \text{ ms} \end{aligned}$$

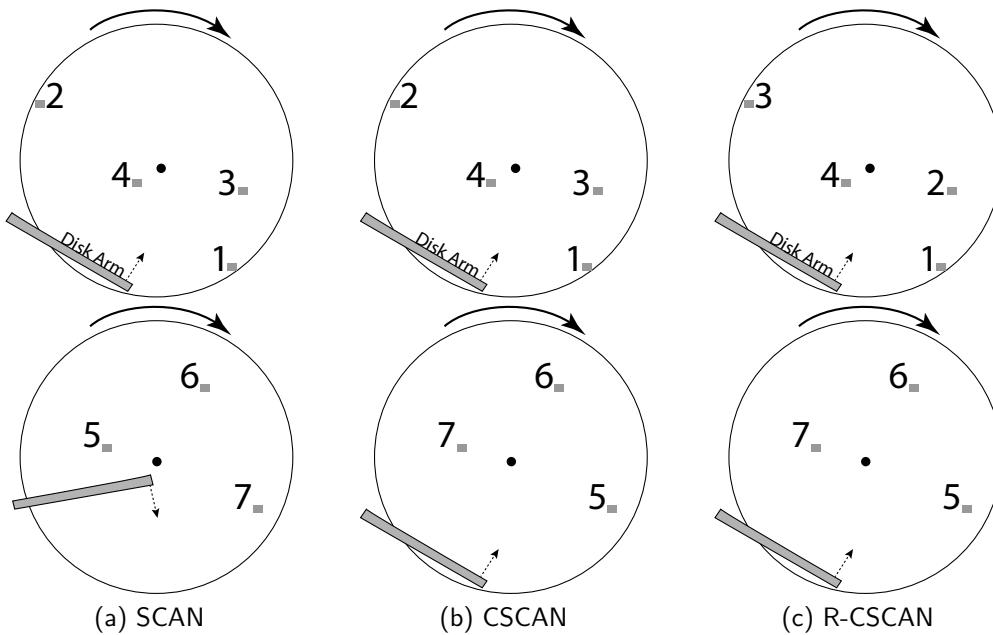


Figure 12.4: Elevator-based scheduling algorithms: (a) SCAN, (b) CSCAN, and (c) R-CSCAN.

Elevator, SCAN, and CSCAN. Elevator-based algorithms like SCAN and CSCAN have good performance and also ensure fairness in that no request is forced to wait for an inordinately long time. The basic approach is similar to how an elevator works: when an elevator is going up, it keeps going up until all pending requests to go to floors above it have been satisfied; then, when an elevator is going down, it keeps going down until all pending requests to go to floors below it have been satisfied.

The *SCAN* scheduler works in the same way. The disk arm first sweeps from the inner to the outer tracks, servicing all requests that are between the arm's current position and the outer edge of the disk. Then, the arm sweeps from the outer to the inner tracks. Then the process is repeated. Figure 12.4-(a) illustrates the SCAN algorithm travelling from outer-to-inner tracks to service four pending requests and then travelling from inner-to-outer tracks to service three additional requests.

Definition: **SCAN**

The *CSCAN* (circular SCAN) scheduler is a slight variation on SCAN in which the disk only services requests when the head is traveling in one direction (e.g., from inner tracks to outer ones); when the last request in the direction of travel is reached, the disk immediately seeks to where it started (e.g., the most inner track or the most inner track with a pending request) and services pending requests by moving the head *in the same direction* as the original pass (e.g., from inner tracks to outer ones again.) Figure 12.4-(b) illustrates the CSCAN algorithm travelling from outer-to-inner tracks to service four pending

Definition: **CSCAN**

requests and then skipping to the outer track and travelling from outer-to-inner tracks to service three additional requests.

The advantage of CSCAN over SCAN is that if after a pass in one direction, the disk head were to just switch directions (as in SCAN), it will encounter a region of the disk where pending requests are sparse (since this region of the disk was just serviced). Seeking to the opposite side of the disk (as in CSCAN) moves the disk head to an area where pending requests are likely to be denser. In addition, CSCAN is more fair than SCAN in that seeking to the opposite side of the disk allows it to begin servicing the requests that likely been waiting longer than requests near but “just behind” the head.

Rather than pure seek-minimizing SCAN or CSCAN, schedulers also take into account rotation time and allow small seeks “in the wrong direction” to avoid extra rotational delays using the rotationally-aware *R-SCAN* or *R-CSCAN*.

Definition: R-SCAN For example, if the disk head is currently over sector 0 of track 0 and there are pending requests at sector 1000 of track 0, sector 500 of track 1, and sector 0 of track 10,000, a R-CSCAN scheduler might service the second request, then the first, and then the third. Figure 12.4-(c) illustrates the R-CSCAN algorithm handling a request on the outer track, then one a few tracks in, then another request on the outer track, and a request near the center on the arm’s first sweep. The arm’s second sweep is the same as for CSCAN.

Example: Effect of disk scheduling.

Question: For the disk described in Figure 12.3, consider a workload consisting of 500 read requests, each of a randomly chosen sector on disk, assuming that the disk head is on the outside track and that requests are serviced in CSCAN order from outside to inside. How long will servicing these requests take?

Answer: Answering a question like this requires making some educated guesses; different people may come up with different reasonable estimates here.

Seek time. We first note that with 500 pending requests spread randomly across the disk, the average seek from one request to the next will seek 0.2% of the way across the disk. With four surfaces, most of these seeks will also require a head switch. We don’t know the exact time for a seek 0.2% of the way across the disk, but we can estimate it by interpolating between the time for a 1 track seek (1 ms) and the time for a 33.3% seek (10.5 ms for reads.) (Disk seek time is not actually linear in distance, but as we will see in a moment, the exact seek time seems unlikely to affect our answer much.)

$$\begin{aligned}\text{estimated .2\% seek time} &= \left(1 + \frac{2}{33.3}\right) 10.5 \text{ ms} \\ &= 1.06 \text{ ms}\end{aligned}$$

Rotation time. Since we don't know the position of the disk when the seek finishes and since sectors are scattered randomly, a simple and reasonable estimate for the time after the seek finishes for the desired block to rotate to the disk head is 4.15 ms, one half of the time that it takes a 7200 RPM disk to rotate once.

Transfer time. Similar to the example on page 361, transfer time for each sector is at most 0.0095 ms

Total time. $1.06 + 4.15 + .0095 = 5.22$ ms per request, so 500 requests will take about 2.6 s. Notice that the time for the SCAN-scheduled time is less than half the 7.8 s time for the FIFO-scheduled time for the example on page 361

Exercises

1. **Discussion.** Some high-end disks in the 1980s had multiple disk arm assemblies per disk enclosure in order to allow them to achieve higher performance. Today, high-performance server disks have a single arm assembly per disk enclosure. Why do you think disks so seldom have multiple disk arm assemblies today?
2. How many sectors does a track on the disk described in Figure 12.3 on page 360 have?
3. For the disk in Figure 12.3 on page 360, estimate the distance from the center of one track to the center of the next track.
4. A disk may have multiple surfaces, arms, and heads, but when you issue a read or write, only one head is active at a time. It seems like one could greatly increase disk bandwidth for large requests by reading or writing with all of the heads at the same time. Given the physical characteristics of disks, can you figure out why no one does this?
5. For the disk described in Figure 12.3 on page 360, consider a workload consisting of 500 read requests, each of a randomly chosen sector on disk, assuming that the disk head is on the outside track and that requests are serviced in P-CSCAN order from outside to inside. How long will servicing these requests take?

Note: Answering this question will require making some estimates.

6. Suppose I have a disk such as the 320GB SATA drive described in Figure 12.9 on page 381 and I have a workload consisting of 10000 reads to sectors randomly scattered across the disk. How long will these 10000 request take (total) assuming the disk services requests in FIFO order?
7. Suppose I have a disk such as the 320GB SATA drive described in Figure 12.9 on page 381 and I have a workload consisting of 10000 reads to 10000 sequential sectors on the outer-most tracks of the disk. How long will these 10000 request take (total) assuming the disk services requests in FIFO order?
8. Suppose I have a disk such as the 320GB SATA drive described in Figure 12.9 on page 381 and I have a workload consisting of 10000 reads to sectors randomly scattered across the disk. How long will these 10000 request take (total) assuming the disk services requests using the SCAN/Elevator algorithm.
9. Suppose I have a disk such as the 320GB SATA drive described in Figure 12.9 on page 381 and I have a workload consisting of 10000 reads to sectors randomly scattered across a 100MB file, where the 100MB file is laid out sequentially on the disk. How long will these 10000 request take (total) assuming the disk services requests using the SCAN/Elevator algorithm?
10. Write a program that creates a 100MB file on your local disk and then measures the time to do each of four things:
 - (a) **Sequential overwrite.** Overwrite the file with 100MB of new data by writing the file from beginning to end and then calling `fsync()` (or the equivalent on your platform).
 - (b) **Random buffered overwrite.** Do the following 50,000 times: choose a 2KB-aligned offset in the file uniformly at random, seek to that location in the file, and write 2KB of data at that position. Then, once all 50,000 writes have been issued, call `fsync()` (or the equivalent on your platform).
 - (c) **Random buffered overwrite.** Do the following 50,000 times: choose a 2KB-aligned offset in the file uniformly at random, seek to that location in the file, write 2KB of data at that position, and call `fsync()` (or the equivalent on your platform) after each individual write.
 - (d) **Random read.** Do the following 50,000 times: choose a 2KB-aligned offset in the file uniformly at random, seek to that location in the file, and read 2KB of data at that position.

Explain your results.

11. Write a program that creates three files, each of 100MB, and then measures the time to do each of three things:
 - (a) **fopen()/fwrite()**. Open the first file using `fopen()` and issue 256,000 sequential four-byte writes using `fwrite()`.
 - (b) **open()/write()**. Open the second file using `open()` and issue 256,000 sequential four-byte writes using `write()`.
 - (c) **mmap()/store**. Map the third file into your program's memory using `mmap()` and issue 256,000 sequential four-byte writes by iterating through memory and writing to each successive word of the mapped file.

Explain your results.

12.2 Flash storage

Over the past decade, flash storage has become a widely used storage medium. Flash storage is the dominant storage technology for handheld devices from phones to cameras to thumb drives, and it is used in an increasing fraction of laptop computers and machine room servers.

Flash storage is a type of *solid state storage*: it has no moving parts and stores data using electrical circuits. Because it has no moving parts, flash storage can have much better random IO performance than disks, and it can use less power and be less vulnerable to physical damage. On the other hand, flash storage remains significantly more expensive per byte of storage than disks.

Definition: solid state storage

Each flash storage element is a floating gate transistor. As Figure 12.5 illustrates, an extra gate in such a transistor “floats”—it is not connected to any circuit. Since the floating gate is entirely surrounded by an insulator, it will hold an electrical charge for months or years without requiring any power. Even though the floating gate is not electrically connected to anything, it can be charged or discharged via electron tunneling by running a sufficiently high-voltage current near it. The floating gate's state of charge affects the transistor's threshold voltage for activation. Thus, the floating gate's state can be detected by applying an intermediate voltage to the transistor's control gate that will only be sufficient to activate the transistor if the floating gate is charged.

In single-level flash storage, the floating gate stores one bit (charge or not charged); in multi-level flash storage, the floating gate stores multiple bits by storing one of several different charge levels.

NOR flash storage is wired to allow individual words to be written and read. NOR flash storage is useful for storing device firmware since it can be executed

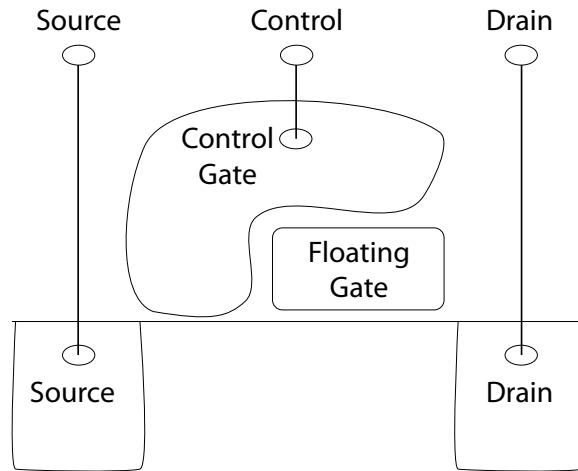


Figure 12.5: A floating gate transistor.

in place. NAND flash storage is wired to allow reads and writes of a page at a time, where a page is typically 2 KB to 4 KB. NAND flash is more dense than NOR flash, so NAND is used in the storage systems we will consider.

Flash storage access and performance Flash storage is accessed using three operations.

Definition: **erasure blocks**

- **Erase erasure block.** Before flash memory can be written, it must be erased by setting each cell to a logical “1”. Flash memory can only be erased in large units called *erasure blocks*. Today, erasure blocks are often 128 KB to 512 KB. Erasure is a slow operation, usually taking several milliseconds.
- Erasing an erasure block is what gives flash memory its name for its resemblance to the flash of a camera.
- **Write page.** Once erased, NAND flash memory can be written on a page by page basis, where each page is typically 2048-4096 bytes. Writing a page typically takes tens of microseconds.
- **Read page.** NAND flash memory can be read on a page by page basis. Reading a page typically takes tens of microseconds.

Notice that to write a page, its entire erasure block must first be erased. This is a challenge both because erasure is slow and because erasure affects a large number of pages. Flash drives implement a *flash translation layer* (FTL) that maps logical flash pages to different physical pages on the flash device. Then, when a single logical page is overwritten, the FTL writes the new version

Definition: **flash translation layer**

to some free, already-erased physical page and remaps the logical page to that physical page.

Write remapping significantly improves flash performance.

Example: Remapping flash writes.

Question: Consider a flash drive with a 4 KB pages, 512 KB erasure blocks, 3 ms flash times, and 50 μ s read-page and write-page times. Suppose writing a page is done with a naive algorithm that reads an entire erasure block, erases it, and writes the modified erasure block. How long would each page write take?

Answer: This naive approach would require

$$\frac{512\text{KB/erasure block}}{4\text{KB/page}} * (\text{page read time} + \text{page write time}) + \text{erasure block erase time} = \\ 128 * (50 + 50)\mu\text{s} + 3\text{ms} = 15.8\text{ms per write.}$$

Question: Suppose remapping is used and that a flash device always has at least one unused erasure block available for a target workload. How long does an average write take now?

Answer: With remapping, the cost of flashing an erasure block is amortized over $512/4 = 128$ page writes. This scenario gives a cost of $\frac{3\text{ms}}{128} + 50\mu\text{s} = 73.4\mu\text{s}$ per write.

In practice, there is likely to be some additional cost per write under the remapping scheme because in order to flash an erasure block to free it for new writes, the firmware may need to garbage collect live pages from that erasure block and copy those live pages to a different erasure block.

Internally, a flash device may have multiple independent data paths that can be accessed in parallel. Therefore, to maximize sustained bandwidth when accessing a flash device, operating systems issue multiple concurrent requests to the device.

Durability. Normally, flash memory can retain its state for months or years without power. However, over time the high current loads from flashing and writing memory causes the circuits to degrade. Eventually, after a few thousand to a few million program-erase cycles (depending on the type of flash), a given cell may *wear out* and no longer reliably store a bit.

Definition: **wear out**

In addition, reading a flash memory cell a large number of times can cause the surrounding cells' charges to be disturbed. A *read disturb error* can occur if a location in flash memory is read to many times without the surrounding memory being written.

Definition: **read disturb error**

To improve durability in the face of wear from writes and disturbs from reads, flash devices make use of a number of techniques:

- **Error correcting codes.** Each page has some extra bytes that are used for error correcting codes to protect against bit errors in the page.
- **Bad page and bad erasure block management.** If a page or erasure block has a manufacturing defect or wears out, firmware on the device marks it as bad and stops storing data on it.
- **Wear leveling.** As noted above, rather than overwrite a page in place, the flash translation layer remaps the logical page to a new physical page that has already been erased. This remapping ensures that a hot page that is overwritten repeatedly does not prematurely wear out a particular physical page on the flash device.

Definition: **wear leveling**

Wear leveling moves a flash device's logical pages to different physical pages to ensure that no physical page gets an inordinant number of writes and wears out prematurely. Some wear leveling algorithms also migrate unmodified pages to protect against read disturb errors.

- **Spare pages and erasure blocks.** Flash devices can be manufactured with spare pages and spare erasure blocks in the device. This spare capacity serves two purposes.

First, it provides extra space for wear leveling: even if the device is logically “full” the wear leveling firmware can copy live pages out of some existing erasure blocks into a spare erasure block, allowing it to flash those existing erasure blocks.

Second, it allows bad page and bad erasure block management to function without causing the logical size of the device to shrink.

In addition to affecting reliability, wear out affects a flash device's performance over time.

First, as a device wears out, accesses may require additional retries, slowing them. Second, as spare pages and erasure blocks are consumed by bad ones, the wear leveling algorithms have less spare space and have to garbage collect live pages—copying them out of their existing erasure blocks—more frequently.

Example: Intel 710 Series Solid-State Drive Figure 12.6 shows some key parameters for an Intel 710 Series solid state drive manufactured in 2011. This drive uses multi-level NAND flash to get high storage densities. Normally, multi-level flash is less durable than single-level, but this Intel drive uses sophisticated wear leveling algorithms and a large amount of spare space to provide high durability.

Size	
Capacity	300 GB
Page Size	4KB
Performance	
Bandwidth (Sequential Reads)	270 MB/s
Bandwidth (Sequential Writes)	210 MB/s
Read/Write Latency	75 μ s
Random Reads Per Second	38,500
Random Writes Per Second	2,000 (2,400 with 20% space reserve)
Interface	SATA 3 Gb/s
Endurance	
Endurance	1.1 PB (1.5 PB with 20% space reserve)
Power	
Power Consumption Active/Idle	3.7 W / 0.7 W

Figure 12.6: Key parameters for an Intel 710 Series Solid State Drive manufactured in 2011.

The sequential performance of this drive is very good, with peak sustained read and write bandwidths of 270 MB/s and 210 MB/s respectively. In comparison, a high-end Seagate Cheetah 15K.7 drive manufactured in 2010 spins at 15,000 revolutions per second and provides 122 MB/s to 204 MB/s of sustained bandwidth.

Random read performance is excellent. The latency for a single random 4 KB read is just 75 μ s, and when multiple concurrent requests are in flight, the drive can process 38,500 random reads per second—one every 26 μ s. This is orders of magnitude better than the random read performance of a spinning disk drive.

Random write performance is also very good, but not as good as random read performance. The latency for a single random 4 KB write is 75 μ s; the drive reduces write latency by buffering writes in volatile memory, and it has capacitors that store enough charge to write all buffered updates to flash storage if a power loss occurs.

When multiple concurrent writes are in flight, the drive can process 2,000 random writes per second when it is full; if it is less than 80% full, that number rises to 2,400. Random write throughput increases when the drive has more free space because the drive has to garbage collect live pages from erasure blocks less often and because when the drive eventually does do that garbage collection, the erasure blocks are less full.

The drive's endurance is rated for 1.1 PB (1.1×10^{15} bytes) of endurance (1.5 PB if it is less than 80% full.) For many workloads, this endurance suffices for years or decades of use. However, solid state drives may not always be a good match for high-bandwidth write streaming. In the extreme, an application constantly

Technology Affects Interfaces—the TRIM Command

Historically, when a file system deleted a file stored on a spinning disk, all it needed to do was to update the file's metadata and the file system's free space bitmap. It did not need to erase or overwrite the file's data blocks on disk—once the metadata was updated, these blocks could never be referenced, so there was no need to do anything with them.

When such file systems were used with flash drives, users observed that their drives got slower over time. As the amount of free space fell, the drives' flash translation layer was forced to garbage collect erasure blocks more frequently; additionally, each garbage collection pass became more expensive because there were more live pages to copy from old erasure blocks to the new ones.

Notice that this slowing could occur even if the file system appeared to have a large amount of free space. For example, if a file system moves a large file from one range of blocks to another, the storage hardware has no way to know that the pages in the old range are no longer needed unless the file system can tell it so.

The TRIM command was introduced into many popular operating systems between 2009 and 2011 to allow file systems to inform the underlying storage when the file system has stopped using a page of storage. The TRIM command makes the free space known to the file system visible to the underlying storage layer, which can significantly reduce garbage collection overheads and help flash drives retain good performance as they age.

streaming writes at 200 MB/s could wear this drive out in 64 days.

Example: Random read workload.

Question: For the solid state disk described in Figure 12.6, consider a workload consisting of 500 read requests, each of a randomly chosen page. How long will servicing these requests take?

Answer: The disk can service random read requests at a rate of 38,500 per second, so 500 requests will take $500/38500 = 13$ ms. In contrast, for the spinning disk example on page 360, the same 500 requests would take 7.8 seconds.

Example: Random v. sequential reads.

Question: How does this drive's random read performance compare to its sequential read performance?

Answer: The effective bandwidth in this case is 500 requests * 4 $\frac{\text{KB}}{\text{request}}$ /

13 milliseconds = 158 MB/s. The random read bandwidth is thus $158/270 = 59\%$ of the sequential read bandwidth.

Example: Random write workload.

Question: For the solid state disk described in Figure 12.6, consider a workload consisting of 500 write requests, each of a randomly chosen page. How long will servicing these requests take?

Answer: The disk can service random write requests at a rate of 2000 per second (assuming the disk is nearly full), so 500 requests will take $500/2000 = 250$ ms.

Example: Random v. sequential writes.

Question: How does this random write performance compare to the drive's sequential write performance?

Answer: The effective bandwidth in this case is $500 \text{ requests} * 4 \frac{\text{KB}}{\text{request}} / 250 \text{ milliseconds} = 8.2 \text{ MB/s}$. The random write bandwidth is thus $8.2/210 = 3.9\%$ of the sequential write bandwidth.

Exercises

12. Suppose that you have a 256 GB solid state drive that the operating system and drive both support the TRIM command. To evaluate the drive, you do an experiment where you time the system's write performance for random page-sized when the file system is empty compared to its performance when the file system holds 255 GB of data, and you find that write performance is significantly worse in the latter case.

What is the likely reason for this worse performance as the disk fills despite its support for TRIM?

What can be done to mitigate this slowdown?

13. Suppose you have a flash drive such as the one described in Figure 12.10 on page 383 and you have a workload consisting of 10000 4KB reads to pages randomly scattered across the drive. Assuming that you wait for request i to finish before you issue request $i + 1$, how long will these 10000 request take (total)?

14. Suppose you have a flash drive such as the one described in Figure 12.10 on page 383 and you have a workload consisting of 10000 4KB reads to pages randomly scattered across the drive. Assuming that you issue requests concurrently, using many threads, how long will these 10000 request take (total)?
15. Suppose you have a flash drive such as the one described in Figure 12.10 on page 383 and you have a workload consisting of 10000 4KB writes to pages randomly scattered across the drive. Assuming that you wait for request i to finish before you issue request $i + 1$, how long will these 10000 requests take (total)?
16. Suppose you have a flash drive such as the one described in Figure 12.10 on page 383 and you have a workload consisting of 10000 4KB writes to pages randomly scattered across the drive. Assuming that use a large number of threads to issue many writes concurrently, how long will these 10000 requests take (total)?
17. Suppose you have a flash drive such as the one described in Figure 12.10 on page 383 and you have a workload consisting of 10000 4KB reads to 10000 sequential pages. How long will these 10000 request take (total)?

12.3 Conclusions and future directions

Today, spinning disk and flash memory dominate storage technologies, and each has sufficient advantages to beat the other for some workloads and environments.

Spinning disk v. flash storage. Spinning disks are often used when capacity is the primary goal. For example, spinning disk is often used for storing media files and home directories. For workloads limited by storage capacity spinning disks can often provide much better capacity per dollar than flash storage. For example, in October 2011, a 2 TB Seagate Barracuda disk targeted at workstations cost about \$80 and a 300 GB Intel 320 Series solid state drive targeted at laptops cost about \$600, giving the spinning disk about a 50:1 advantage in GB per dollar.

Both spinning disks and flash storage are viable when sequential bandwidth is the goal. In October 2011, flash drives typically have modestly higher per-drive sequential bandwidths than spinning drives, but the spinning drives typically have better sequential bandwidth per dollar spent than flash drives. For example, the same Seagate disk has a sustained bandwidth of 120 MB/s (1.5



Figure 12.7: In 2011, flash storage “keys” such as this one can store as much as 256 GB in a device that is a few centimeters long, and 1-2 cm wide and tall.

MB/s per dollar) while the same Intel SSD has a read/write bandwidth of 270/205 MB/s (about 0.4 MB/s per dollar.)

Flash storage is often used when good random access performance or low power consumption is the goal. For example, flash storage is frequently used in database transaction processing servers, in smart phones, and in laptops. For example, the Seagate described above drive rotates at 5900 RPM, so it takes about 5 ms for a half rotation; even with good scheduling and even if data is confined to a subset of tracks, it would be hard to get more than 200 random I/Os per second from this drive (about 2.5 random I/Os per second per dollar.) Conversely, the Intel SSD can sustain 23,000 random writes and 39,500 random reads per second (about 38 or 66 random writes or reads per second per dollar.)

With respect to power, spinning disks typically consume 10-20W depending on whether it is just spinning or actively reading and writing data, while a flash drive might consume 0.5W-1W when idle and 3-5W when being accessed. Flash drives’ power advantage makes them attractive for portable applications such as laptop and smartphone storage.

Flash memory can also have a significant form factor advantage with respect to physical size and weight. Although some flash drives are designed as drop-in replacements for spinning disks and so are similar in size, flash storage can be much smaller than a typical spinning disk. For example, in 2011, a USB flash storage “key” such as the one in Figure 12.7 can store as much as 256 GB in a device that is not much larger than a house key.

Figure 12.8 summarizes these advantages and disadvantages; of course, many systems need to do well on multiple metrics, so system designers may need to compromise on some metrics or use combinations of technologies.

Technology trends. Over the past decades, the cost of storage capacity has fallen rapidly for both spinning disks and solid state storage. Compare the 2 TB disk drive for \$80 in 2011 to a 15 MB drive costing \$113 in 1984 (or about \$246

Metric	Spinning Disk	Flash
Capacity/Cost	Excellent	Good
Sequential BW/Cost	Good	Good
Random I/O per Second/Cost	Poor	Good
Power Consumption	Fair	Good
Physical Size	Good	Excellent

Figure 12.8: Relative advantages and disadvantages of spinning disk and flash storage.

in 2011 dollars): the cost per byte has improved by a factor of about 400,000 over 27 years—over 50% per year for nearly 3 decades.

Recent rates of improvement for flash storage have been even faster. For example, in 2001, the Adtron S35PC 14 GB flash drive cost \$42,000. Today’s Intel 320 costs 70 times less for 21 times more capacity, an improvement of about 2x per year over the past decade.

Similar capacity improvements for spinning disk and flash are expected for at least the next few years. Beyond that, there is concern that we will be approaching the physical limits of both magnetic disk and flash storage, so the longer-term future is less certain. (That said, people have worried that disks were approaching their limits several times in the past, and we will not be surprised if the magnetic disk and flash industries continue rapid improvements for quite a few more years.)

In contrast to capacity, performance is likely to improve more slowly for both technologies. For example, a mid-range spinning disk in 1991 might have had a 1.3 MB/s maximum bandwidth and an 17 ms average seek time. Bandwidths have improved by about a factor of 90 in two decades (about 25% per year) while seek times and rotational latencies have only improved by about a factor of two (less than 4% per year.) Bandwidths have improved more quickly than rotational latency and seek times because bandwidth benefits from increasing storage densities, not just increasing rotational rates.

For SSDs, the story is similar, though recent increases in volumes have helped speed the pace of improvements. For example, in 2006 a BitMicro E-Disk flash drive could provide 9,500 to 11,700 random reads per second and 34-44 MB/s sustained bandwidth. Compared to the Intel 320 SSD from 2011, bandwidths have improved by about 40% per year and random access throughput has improved by about 25% per year over the past 5 years.

New technologies. This is an exciting time for persistent storage. After decades of undisputed reign as the dominant technology for on-line persistent storage, spinning magnetic disks are being displaced flash storage in many

The first disk drive

Prior to the invention of magnetic disks, magnetic cylinders, called drums, were used for on-line storage. These drums spun on their axes and typically had one head per track. So, there was no seek time to access a block of data; one merely waited for a block to rotate underneath its head.

By using spinning disks instead of drums, the magnetic surface area, and hence the storage capacity, could be increased.



Photo by U. S. Army Red River Arsenal.

The first disk drive, the IBM 350 Disk System (two are shown in the foreground of this photograph), was introduced in 1956 as part of the IBM RAMAC ("Random Access Method of Accounting and Control") 305 computer system. The 350 Disk system stored about 3.3 MB on 50 platters, rotated its platters at 1200 RPM, had an average seek time of 600 ms, and weighed about a ton. The RAMAC 305 computer system with its 350 disk system could be leased for \$3,200 per month. Assuming a useful life of 5 years and converting to 2011 dollars, the cost was approximately \$1.3 million for the system—about \$400,000 per megabyte.

application domains, giving both operating system designers and application writers an opportunity to reexamine how to best use storage. Looking forward, many researchers speculate that new technologies may soon be nipping at the heels and even surpassing flash storage.

Definition: **phase change memory**

For example *phase change memory* (PCM) uses a current to alter the state of chalcogenide glass between amorphous and crystalline forms, which have significantly different electrical resistance and can therefore be used to represent data bits. Although PCM does not yet match the density of flash, researchers speculate that the technology is fundamentally more scalable and will ultimately be able to provide higher storage densities at lower costs. Furthermore, PCM is expected to have much better write performance and endurance than flash.

Definition: **memristors**

As another example, *memristors* are circuit elements whose resistance depends on the amounts and directions of currents that have flowed through them in the past. A number of different memristor constructions are being pursued, and some have quite promising properties. For example, in 2010 Hewlett Packard labs described a prototype memristor constructed of a thin titanium dioxide film with 3 nm by 3 nm storage elements that can switch states in 1 ns. These densities are similar to contemporary flash memory devices and these switching times are similar to contemporary DRAM chips. The devices also have write endurance similar to flash, and extremely long (theoretically unlimited) storage lifetimes. Furthermore, researchers believe that these and others memristors' densities will scale well in the future. For example, in 2009 a design for 3-D stacking of memristors was published in the *Proceedings of National Academy of Sciences* by Dmitri Strukov and R. Stanley Williams of HP Labs.

If technologies such as these pan out as hoped, operating system designers will have opportunities to rethink our abstractions for both volatile and nonvolatile storage: how should we make use of word-addressable, persistent memory with densities exceeding current flash storage devices and with memory access times approaching those of DRAM? What could we do if each core on a 32 core processor chip had access to a few gigabytes of stacked memristor memory?

Exercises

1. **Discussion.** Some high-end disks in the 1980s had multiple disk arm assemblies per disk enclosure in order to allow them to achieve higher performance. Today, high-performance server disks have a single arm assembly per disk enclosure. Why do you think disks so seldom have multiple disk arm assemblies today?
2. How many sectors does a track on the disk described in Figure 12.3 on page 360 have?

Size	
Form factor	2.5-inch
Capacity	320 GB
Performance	
Spindle speed	5400 RPM
Average seek time	12.0 ms
Maximum seek time	21 ms
Track-to-track seek time	2 ms
Transfer rate (surface to buffer)	850 Mbit/s (maximum)
Transfer rate (buffer to host)	3 Gbit/s
Buffer memory	8 MB

Figure 12.9: Hardware specifications for a 320GB SATA disk drive.

3. For the disk in Figure 12.3 on page 360, estimate the distance from the center of one track to the center of the next track.
4. A disk may have multiple surfaces, arms, and heads, but when you issue a read or write, only one head is active at a time. It seems like one could greatly increase disk bandwidth for large requests by reading or writing with all of the heads at the same time. Given the physical characteristics of disks, can you figure out why no one does this?
5. For the disk described in Figure 12.3 on page 360, consider a workload consisting of 500 read requests, each of a randomly chosen sector on disk, assuming that the disk head is on the outside track and that requests are serviced in P-CSCAN order from outside to inside. How long will servicing these requests take?

Note: Answering this question will require making some estimates.

6. Suppose I have a disk such as the 320GB SATA drive described in Figure 12.9 on page 381 and I have a workload consisting of 10000 reads to sectors randomly scattered across the disk. How long will these 10000 request take (total) assuming the disk services requests in FIFO order?
7. Suppose I have a disk such as the 320GB SATA drive described in Figure 12.9 on page 381 and I have a workload consisting of 10000 reads to 10000 sequential sectors on the outer-most tracks of the disk. How long will these 10000 request take (total) assuming the disk services requests in FIFO order?
8. Suppose I have a disk such as the 320GB SATA drive described in Figure 12.9 on page 381 and I have a workload consisting of 10000 reads to

- sectors randomly scattered across the disk. How long will these 10000 request take (total) assuming the disk services requests using the SCAN/Elevator algorithm.
9. Suppose I have a disk such as the 320GB SATA drive described in Figure 12.9 on page 381 and I have a workload consisting of 10000 reads to sectors randomly scattered across a 100MB file, where the 100MB file is laid out sequentially on the disk. How long will these 10000 request take (total) assuming the disk services requests using the SCAN/Elevator algorithm?
 10. Write a program that creates a 100MB file on your local disk and then measures the time to do each of four things:
 - (a) **Sequential overwrite.** Overwrite the file with 100MB of new data by writing the file from beginning to end and then calling `fsync()` (or the equivalent on your platform).
 - (b) **Random buffered overwrite.** Do the following 50,000 times: choose a 2KB-aligned offset in the file uniformly at random, seek to that location in the file, and write 2KB of data at that position. Then, once all 50,000 writes have been issued, call `fsync()` (or the equivalent on your platform).
 - (c) **Random buffered overwrite.** Do the following 50,000 times: choose a 2KB-aligned offset in the file uniformly at random, seek to that location in the file, write 2KB of data at that position, and call `fsync()` (or the equivalent on your platform) after each individual write.
 - (d) **Random read.** Do the following 50,000 times: choose a 2KB-aligned offset in the file uniformly at random, seek to that location in the file, and read 2KB of data at that position.
 - Explain your results.
 11. Write a program that creates three files, each of 100MB, and then measures the time to do each of three things:
 - (a) **`fopen()`/`fwrite()`.** Open the first file using `fopen()` and issue 256,000 sequential four-byte writes using `fwrite()`.
 - (b) **`open()`/`write()`.** Open the second file using `open()` and issue 256,000 sequential four-byte writes using `write()`.
 - (c) **`mmap()`/`store`.** Map the third file into your program's memory using `mmap()` and issue 256,000 sequential four-byte writes by iterating through memory and writing to each successive word of the mapped file.

Size	
Usable capacity	2 TB (SLC flash)
Cache Size	64 GB (Battery-backed RAM)
Page Size	4KB
Performance	
Bandwidth (Sequential Reads from flash)	2048 MB/s
Bandwidth (Sequential Writes to flash)	2048 MB/s
Read Latency (cache hit)	15 μ s
Read Latency (cache miss)	200 μ s
Write Latency	15 μ s
Random Reads (sustained from flash)	100,000 per second
Random Writes (sustained to flash)	100,000 per second
Interface	8 Fibre Channel ports with 4Gbit/s per port
Power	
Power Consumption	300 W

Figure 12.10: Key parameters for a hypothetical high-end flash drive in 2011.

Explain your results.

12. Suppose that you have a 256 GB solid state drive that the operating system and drive both support the TRIM command. To evaluate the drive, you do an experiment where you time the system's write performance for random page-sized when the file system is empty compared to its performance when the file system holds 255 GB of data, and you find that write performance is significantly worse in the latter case.

What is the likely reason for this worse performance as the disk fills despite its support for TRIM?

What can be done to mitigate this slowdown?

13. Suppose you have a flash drive such as the one described in Figure 12.10 on page 383 and you have a workload consisting of 10000 4KB reads to pages randomly scattered across the drive. Assuming that you wait for request i to finish before you issue request $i + 1$, how long will these 10000 request take (total)?

14. Suppose you have a flash drive such as the one described in Figure 12.10 on page 383 and you have a workload consisting of 10000 4KB reads to pages randomly scattered across the drive. Assuming that you issue requests concurrently, using many threads, how long will these 10000 request take (total)?

15. Suppose you have a flash drive such as the one described in Figure 12.10 on page 383 and you have a workload consisting of 10000 4KB writes to pages randomly scattered across the drive. Assuming that you wait for request i to finish before you issue request $i + 1$, how long will these 10000 requests take (total)?
16. Suppose you have a flash drive such as the one described in Figure 12.10 on page 383 and you have a workload consisting of 10000 4KB writes to pages randomly scattered across the drive. Assuming that use a large number of threads to issue many writes concurrently, how long will these 10000 requests take (total)?
17. Suppose you have a flash drive such as the one described in Figure 12.10 on page 383 and you have a workload consisting of 10000 4KB reads to 10000 sequential pages. How long will these 10000 request take (total)?

Chapter 13

Files and Directories

*What's in a name? That which we call a rose
By any other name would smell as sweet.*

— Juliet

Romeo and Juliet (II, ii, 1-2)
(Shakespeare)

File Systems

```
-- Intro: list of "wants" (excluding reliable) v. challenges v. fs
technique

key idea: file = name for bucket of data

name -----> file num -----> file blocks
          directory           file

-- File

abstraction file num -> metadata, blocks
  metadata -- size, creation time, owner, security permissions,
            ...
            ...

-- basic abstractions

flat file system: (names come later)
  open, close, read, write
```

```
define each one
walk through each one

many different organizations, but all must solve same problems

basic goal: need a map from <file num, blocknum> --> block
on disk

free space

locality heuristics

--- FAT: Linked list
--- FFS: Fixed tree
--- NTFS: flexible tree
--- WAFS: Write anywhere file systems
(e.g., WAFL, LFS,

-- Directories

-- SEcurity

-- Alternatives

-- Future...
```

13.1 Accessing files: API and caching

```
open file, read/write/seek, close file

file names: hierarchical directories
root directory, subdirectories
example: /bin/ls
          /home/mike

current working directory
e.g., cd /home/mike
      ls
      cd

current directory .
parent directory ..
```

e.g., du -s .
e.g., cd ..

Access control

each file -- metadata -- e.g., owner, group, permissions
(read/write/execute) for owner/group/all

stored with file

checked on open/close

open/close:

-> cache file lookup
name resolution (path, current directory)
access control checks (open mode -- read/write/create/delete)
file descriptors/handles

create/delete

read/write/seek
current position

flush

example: create file, write hello world to it,

mapped file API

mmap/m munmap
load/store

example

mmap file
write hello world to it
close/munmap

open file
read and print

DEFER: Sidebar: everything is a file
file handle = level of indirection
a thing you can write to or read from

```
--> can use same API to read/write keyboard/terminal as file  
e.g., cat /etc/motd | echo  
e.g. [code -- open file, open tty, write, write...]
```

13.2 Files: Placing and finding data

file: set of logically related data; often accessed together
e.g., word processor document, object file, music file
not just "data", programs too: word processor, compiler,
mp3 player

flat file system:
you can think of this section as defining a "flat file system"
open/close/read write/seek

need: need FID->blocks of file
additional goals: generic file system (for wide range of applications)
-- excellent sequential access
-- acceptable random access
-- efficient for both large and small files

to meet these goals -- basic idea:
index structure -- head + pointers to rest of dat
e.g., linked list [picture], tree [picture]

free block list -- to maximize sequential layout
e.g., bitmap

allocation policy/defragmentation -- to maximize sequential
layout
e.g., attempt to lay out each file sequentially

Range of ways to do each of these three things. To understand design space, 4 case studies: FAT, FFS, NTFS, WAIFS
-- for each one: index structures, free space, allocation policy

-- progression -- simple/primitive/old to more recent/sophisticated
(still, basic principles fundamentally the same...)

-- all are widely used today

before launching discussion, note
one key issue deferred -- reliability/crash recovery. Forward reference.

13.2.1 FAT: Linked list

index structures
PICTURE
basic data structure and access

FAT -- file allocation table
linked allocation

free space

defragmentation
mechanism v. policy -- mechanism allows any possible layout; policy -- use heuristics to get good layouts
defragmentation

evaluation
simple
sequential access: poor to good
random access: poor to not horrible

discussion/sidebar: FAT is primitive, but widely used
usb sticks
.doc

...

13.2.2 FFS: Fixed tree

(many unix/linux variants; modern examples: ext*)

fixed assymmetric tree

rationale:

tree -> good sequential, random access
tree --> support for sparse files
assymmetric --> small v. large files

index structures:

[picture]

inode array (file ID -> inode)

inode -- metadata

 direct pointers, indirect pointers, double
 indirect, triple indirect

[single/double/triple] indirect blocks

free space: bitmap

locality heuristics: 10% free, cylindar group placement

13.2.3 NTFS: Flexible tree with extents

\mike{Good reference: <http://www.ntfs.com/>}

\mike{Good reference -- low level details of ntfs}

\url{[http://sourceforge.net/projects/linux-ntfs/files/NTFS Documentation/](http://sourceforge.net/projects/linux-ntfs/files/NTFS%20Documentation/)}

additional info:

<http://pages.cs.wisc.edu/~swift/classes/cs537-fa07/lectures/17-ntfs.pdf>

we'll talk about NTFS, but similar "flexible tree with extents" approach in other systems (e.g., XFS, Demos, ...)

flexible tree -- number of layers can vary
rather than one ptr per block, ptr to
variable-range extents

rationale:

extents: don't bother to optimize/allow any possible layout. Large files need sequential layout or you're dead.

--> assume data stored in reasonably large sequential extents

flexible: shallow for small files, large file with few extents
-- allow deeper when needed

index structures:

FIGURE [need to think about what to show here and what to defer for a few paragraphs when talking about specific file layouts...]

maybe: MFT figure
tiny file figure/MFT attributes
med file figure/MFT attributes...

MFT -- array of MFT records

MFT record

series of name/attribute pairs
name: e.g., filename, timestamp, security descriptor, [unamed data]

attribute: stream of bytes within file

--> NTFS doesn't read/write files, it reads/writes attribute streams
-- unifies "metadata" and "data" storage
e.g., space in MFT record for small

```
attribute streams --> store metadata in MFT record; but a  
contents of file with small data  
  
-- extensible -- can add new metadata to  
file  
  
--> unlike FFS inode (fixed structure)  
MFT record has extensible/variable structure  
  
first record: attribute list --> can  
parse file  
[attribute name -> MFT record in  
which attribute is located (for multi-MFT files)  
  
-- supports multi-fork file (NTFS calls  
this X)  
  
-- resident and nonresident attributes  
  
-- example: small file: resident  
-- med-large file -- 1 level  
-- huge file -- 2 level  
...  
  
free space: bitmap file  
  
locality heuristics: ???
```

13.2.4 Write anywhere file systems

```
basic idea: index structures --> file blocks can be stored  
anywhere  
--> when block is updated, don't need to seek and  
overwrite in place; instead can write block  
on any free block near disk head.
```

WAFL does this and goes one better; not only are file
blocks mobile, so are essentially all of the metadata
(index blocks,

why is this good? goals/motivation:

```
transform random writes into sequential
(motivation: caches reduce reads, but still need to
write; bandwidth v. seek gap wide and growing)
allow snapshots

wear leveling (for flash)

additional goal: allow atomic multi-sector update
transactions and reduce crash recovery time; (defer
discussing this until section X)

increasingly widely used
WAFL, LFS, ZFS, FLASH device drivers

implementation: index structures, reads, writes

figure fixed inode --> mobile inode

figure -- updating a file (like Sun slide figure)

mobile metadata: solve "no update in place" problem
level of indirection
better?
more levels of indirection...checkpoint/uberblock

snapshots

defragmentation/cleaning
```

13.2.5 Exercises

QUESTION: linked list, tree. Why not hash table?

QUESTION: spatial locality v. temporal locality

13.3 Directories: Naming data

```
flat file system is a start, but we want names
persistence, sharing --> need human-sensible names
```

```
solution: directories
directory: name->file number

You would invent directories if they didn't already exist
piece of paper...

store in a file
regular file
restrict write interface to maintain structure

hierarchy

example: /foo/bar/baz
example: /foo/bar/baz on FFS

links -- hard and soft

NTFS, ZFS -- special case; ot "just files"

file:///Users/dahlin/Downloads/ntfsdoc-0.5/concepts/directory.html
also -- NTFS directory index entries include
copies of all standard attributes [[so file needs pointer back
to directory to allow atomic update? What about hard links?]]
```

13.4 Putting it all together: File access in FFS

what disk accesses occur when reading first block of /foo/bar/baz
exercise: what updates if create new file /foo/bar/new

13.5 Alternatives to file systems

defer?

quick overview of database
bigtable

[both are key->value stores; no directories; different locality]

```
    patterns;
database tree + log is analogous to
traditional file system; bigtable hashtable is analgous to WAFS]
```

Chapter 14

Reliable Storage

A stitch in time saves nine

– English Proverb

Highly reliable storage is vitally important across a wide range of applications from businesses that need to know that their billing records are safe to families that have photo albums they would like to last for generations.

So far, we have treated disks and flash as ideal nonvolatile storage: data stored there will remain there forever, until it is overwritten. Physical devices cannot achieve such perfection—they may be defective, they may wear out, or they may be damaged—and lose some or all of their data.

Unfortunately, the limits of physical devices are not merely abstract concerns. For example, some large organizations have observed annual disk failure rates of 2% to 4%, meaning that an organization with 10,000 disks might expect to see hundreds of failures per year and that important data stored on a single disk by a naive storage system might have more than a 30% chance of disappearing within a decade.

The central question of this chapter is: How can we make a storage system more reliable than the physical devices out of which they are built?

A system is *reliable* if it performs its intended function. Reliability is related to, but different than, availability. A system is *available* if it currently can respond to a request.

In the case of a storage system, the storage system is reliable as long as it continues to store a given piece of data and as long as its components are capable of reading or overwriting that data. We define a storage system's *reliability* as

the probability that it will continue to be reliable for some specified period of time. A storage system is available at some moment of time if a read or write operation could be completed at that time, and we define a storage system's **availability** as the probability that the system will be available at any given moment of time.

To see the difference between reliability and availability, consider the highly reliable but highly unavailable storage device shown in Figure 14.1. In the 70's, the two Voyager spacecraft sent out of our solar system each included a golden record on which various greetings, diagrams, pictures, natural sounds, and music were encoded as "a present from a small, distant world, a token of our sounds, our science, our images, our music, our thoughts and our feelings." (President Carter) To protect against erosion, the record is encased in an aluminium and uranium cover. This storage device is highly reliable—it is expected to last for many tens of thousands of years in interstellar space—but it is not highly available (at least, not to us.)

To take a more pedestrian example, suppose a storage system required each data block to be written to a disk on each of 100 different machines physically distributed across 100 different machine rooms spread across the world. Such a system might be highly reliable (since it would take a fairly spectacular catastrophe to wipe out all of the copies of any data that is stored), highly available for reads (since there are 100 different locations to read from), but not highly available for writes (since the unavailability of any one of the 100 machines would prevent new writes from completing.)

Two problems. Broadly speaking, storage systems must deal with two threats to reliability.

- **Operation interruption.** A crash or power failure in the middle of a series of related updates may leave the stored data in an inconsistent state. For example, suppose that a user has asked an operating system to move a file from one directory to another:

```
> mv drafts/really-important.doc final/really-important.doc
```

As we discuss in later chapters, such a move may entail many low level operations such as writing the `drafts` directory file to remove `really-important.doc`, updating the last-modified time of the `drafts` directory, growing the `final` directory's file to include another block of storage to accommodate a new directory entry for `really-important.doc`, writing the new directory entry to the directory file, updating the file system's free space bitmap to note that the newly allocated block is now in use, and updating the size and last-modified time of the `final` directory.

Suppose that the system's power fails when the updates to the `drafts` directory are stored in nonvolatile storage but when the updates to the



Figure 14.1: The Voyager “Golden Record,” a highly reliable but highly unavailable storage device. Photo Credit: NASA.

`final` directory are not; in that case, the file `really-important.doc` may be lost. Or, suppose that the operating system crashes after updating the `drafts` and `final` directories but before updating the file system's free space bitmap; in that case, the file system will still regard the new block in the `final` directory as free, and it may allocate that block to be part of some other file. The storage device then ends up with a block that belongs to two files, and updates intended for one file may corrupt the contents of the other file.

- **Loss of stored data.** Failures of nonvolatile storage media can cause previously stored data to disappear or be corrupted. Such failures can affect individual blocks, entire storage devices, or even groups of storage devices.

For example, a disk sector may be lost if it is scratched by a particle contaminating the drive enclosure, a flash memory cell might lose its contents when large numbers of reads of nearby cells disturb its charge, a disk drive can fail completely because bearing wear causes the platters to vibrate too much to be successfully read or written, or a set of drives might be lost when a fire in a data center destroys a rack of storage servers.

Two solutions. Fortunately, system designers have developed two sets of powerful solutions to these problems, and the rest of the chapter discusses them.

- **Transactions for atomic updates.** When a system needs to make several related updates to nonvolatile storage, it may want to ensure that the state is modified atomically: even if a crash occurs the state reflects either all of the updates or none of them. Transactions are a fundamental technique to provide atomic updates of nonvolatile storage

Transactions are simple to implement and to use, and they often have as good or better performance than ad-hoc approaches. The vast majority of widely-used file systems developed over the past two decades have used transactions internally, and many applications implement transactions of their own to keep their persistent state consistent.

- **Redundancy for media failures.** To cope with data loss and corruption, storage systems use several forms of redundancy such as checksums to detect corrupted storage and replicated storage to recover from lost or corrupted sectors or disks.

Implementing sufficient redundancy at acceptably low cost can be complex. For example, a widely-used, simple model of RAID (redundant array of inexpensive disk) paints an optimistic picture of reliability that can be off by orders of magnitude. Modern storage systems often make use of multiple levels of checksums (e.g., both in storage device hardware and file system software), sufficient redundancy to survive two or more hardware failures (e.g., keeping 3 copies of a file or two parity disks with a

RAID), and rely on software that to detect failures soon after they occur and to repair failures quickly (e.g., background processes that regularly attempt to read all stored data and algorithms that parallelize recovery when a device fails.) Systems that fail to properly use these techniques may be significantly less reliable than expected.

14.1 Transactions: Atomic updates

When a system needs to make several updates to nonvolatile storage, and a crash occurs, some of those updates may be stored and survive the crash and others may not. Because a crash may occur without warning, storage systems and applications need to be constructed so that no matter when the crash occurs, the system's nonvolatile storage is left in some sensible state.

This problem occurs in many contexts. For example, if a crash occurs while you are installing an update for a suite of applications, upon recovery you would like to be able to use either the old version or the new version, not be confronted with a mishmash of incompatible programs. For example, if you are moving a subdirectory from one location to another when a crash occurs, when you recover you want to see the data in one location or the other; if the subdirectory disappears because of an untimely crash, you will be (justifiably) upset with the operating system designer. Finally, if a bank is moving \$100 from Alice's account to Bob's account when a crash occurs, it wants to be certain that upon recovery either the funds are in Alice's account and records show that the transfer is still to be done or that the funds are in Bob's account and the records show that the transfer has occurred.

This problem is quite similar to the critical section problem in concurrency. In both cases, we have several updates to make and we want to avoid having anyone observe the state in an intermediate, inconsistent state. Also, we have no control when other threads might try to access the state in the first case or when a crash might occur in the second—we must develop a structured solution that works for any possible execution. The solution is similar, too; we want to make the set of updates atomic. But, because we are dealing with nonvolatile storage rather than main memory, the techniques for achieving atomicity differ in some significant ways.

Transactions extend the concept of atomic updates from memory to stable storage, allowing systems to atomically update multiple persistent data structures.

14.1.1 Ad hoc approaches

Until the mid 1990's many file systems used ad hoc approaches to solving the problem of consistently updating multiple on-disk data structures.

For example, the Unix fast file system (FFS) would carefully control the order that its updates were sent to disk so that if a crash occurred in the middle of a group of updates, a scan of the disk during recovery could identify and repair inconsistent data structures. For example, when creating a new file, FFS would first update the free-inode bitmap to indicate which previously free inode was now in use; after making sure this update was on disk, it would initialize the new file's inode, clearing all of the direct, indirect, double-indirect, and other pointers, setting the file length to 0, setting the file's ownership and access control list, and so on. Finally, once the inode update was safely on disk, the file system would update the directory to contain an entry for the newly created file, mapping the file's name to its inode.

If a system running FFS crashed, then when it rebooted it would use a program called `fsck` to scan all of the file system's metadata (e.g., all inodes, all directories, and all free space bitmaps) to make sure that all metadata items were consistent. For example, if `fsck` discovered an inode that was marked as *allocated* in the free-inode bitmap but that did not appear in any directory entry, it could infer that that inode was part of a file in the process of being created (or deleted) when the crash occurred; since the create had not finished or the delete had started, `fsck` could mark the inode as free, undoing the partially completed create (or completing the partially completed delete.)

Similar logic was used for other file system operations.

This approach of careful ordering of operations with scanning and repair of on-disk data structures was widespread up until the 1990's, when it was largely abandoned. In particular, this approach has three significant problems:

1. **Complex reasoning.** Similar to trying to solve the multi-threaded synchronization problem with just atomic loads and stores, this approach requires reasoning carefully about all possible operations and all possible failure scenarios to make sure that it is always possible to recover the system to a consistent state.
2. **Slow updates.** To ensure that updates are stored in an order that allowed the system's state to be analyzed, file systems are forced to insert `sync` operations or barriers between dependent operations, reducing the amount of pipelining or parallelism in the stream of requests to storage devices.

For example, in the file creation example above to ensure that the individual updates hit disk in the required order, the system might suffer three full rotations of the disk to update three on-disk data structures even though those data structures may be quite near each other.

3. **Extremely slow recovery.** When a machine reboots after a crash, it has to scan all of its disks for inconsistent metadata structures.

In the 1970's and 1980s, it was possible to scan the data structures on a disk in a few seconds or a few minutes, but by the 1990's this scanning

fsync Still Lives

Although few file systems today rely on scanning disks when recovering from a crash, `fsck` (and variants on other operating systems) is still provided as an “emergency fix” when on-disk data structures are corrupted for other reasons (e.g., due to software bug or storage device failure).

could take tens of minutes to a few hours for large servers with many disks, and technology trends indicated that scan times would grow rapidly worse.

Although the first two were significant disadvantages of the approach, it was the third that finally made depending on careful ordering and `fsck` untenable for most file systems. New file systems created since the late 1980’s almost invariably use other techniques—primarily various forms of transactions that we discuss in the rest of this section.

Application-level approaches. Although modern file systems often use transactions internally, some standard file system APIs such as the Posix API provide only weaker abstractions, forcing applications to take their own measures if they want to atomically apply a set of updates. Many use application-level transactions, but some continue to use more ad hoc approaches.

For example, suppose that a user has edited several parts of a text file and then wants to save the updated document. The edits may have inserted text at various points in the document, removed text at others, and shifted the remaining text forwards or backwards—even a small insertion or deletion early in the document could ripple through the rest of the file.

If the text editor application were simply to use the updated file in its memory to overwrite the existing file, an untimely crash could leave the file in an incomprehensible state—the operating system and disk schedulers may choose any order to send the updated blocks to nonvolatile storage, so after the crash the file may be an arbitrary mix of old and new blocks, sometimes repeating sections of text, sometimes omitting them entirely.

To avoid this problem, the text editor may take advantage of the semantics of the Posix `rename` operation, which renames the file called `sourceName` to be called `targetName` instead. Posix promises that if a file named `targetName` already exists, `rename`’s shift from having `targetName` refer to the old file to having it refer to the new one will be atomic. (This atomicity guarantee may be provided by transactions within the file system or by ad hoc means.)

So, to update an existing file `design.txt`, the text editor first writes the

updates to a new, temporary file such as `#design.txt#`. Then it renames the temporary file to atomically replace the previously stored file.

14.1.2 The transaction abstraction

Transactions provide a way to atomically update multiple pieces of persistent state.

For example, suppose you are updating a web site and you want to replace the current collection of documents in `/server/live` with a new collection of documents you have created in `/development/ready`. You don't want users to see intermediate steps when some of the documents have been updated and others have not—they might encounter broken links or encounter new descriptions referencing old pages or vice versa. Transactional file systems like Windows Vista's TxF (Transactional NTFS) provide an API that lets applications apply all of these updates atomically, allowing the programmer to write something like the following pseudo-code:

```
ResultCode publish(){
    transactionID = beginTransaction();
    foreach file f in /development/ready that is not in /server/live{
        error = move f from /development/ready to /server/live;
        if(error){
            rollbackTransaction(transactionID);
            return ROLLED_BACK;
        }
    }

    foreach file f in /server/live that is not in /development/ready{
        error = delete f;
        if(error){
            rollbackTransaction(transactionID);
            return ROLLED_BACK;
        }
    }

    foreach file f in /development/ready that is different than in /server/live{
        error = move f from /development/ready to /server/live;
        if(error){
            rollbackTransaction(transactionID);
            return ROLLED_BACK;
        }
    }
    commitTransaction(transactionID);
    return COMMITTED;
}
```

Definition: **commit** Notice that a transaction can finish in one of two ways: it can *commit*,
 Definition: **roll back** meaning all of its updates occur, or it can *roll back* meaning that none of its
 updates occur.

Here, if the transaction commits, we are guaranteed that all of the updates will be seen by all subsequent reads, but if it encounters an error and rolls back or crashes without committing or rolling back, no reads will see any of the updates.

Definition: **transaction** More precisely, a *transaction* is a way to perform a set of updates while

providing the following ACID properties:

- **Atomicity.** Updates are “all or nothing.” If the transaction *commits*, all updates in the transaction take effect. If the transaction *rolls back*, then none of the updates in the transaction have any effect.

In the website update example above, doing the updates within a transaction guarantees that each of the update is only stored or readable if all of the updates are stored and readable.

- **Consistency.** The transaction moves the system from one legal state to another. A system’s invariants on its state can be assumed to hold at the start of a transaction and must hold when the transaction commits.

In the example above, by using a transaction we can maintain the invariant that every link from one document to another on the server references a valid file.

- **Isolation.** Each transaction appears to execute on its own, and is not affected by other in-progress transactions. Even if multiple transactions execute concurrently, for each pair of transactions T and T' , it either appears that T executed entirely before T' or vice versa.

By executing the web site update in a transaction, we guarantee that each transaction to read from the web site is applied either against the old set of web pages or the new set, not some mix of the two.

Of course, if each individual read of an object is in its own transaction, then a series of reads to assemble a web page and its included elements could see the old web page and a mix of old and new elements. If web protocols were changed to allow a browser to fetch a page and its elements in a single transaction, then we could guarantee that the user would see either the old page and elements or the new ones.

- **Durability.** A committed transaction’s changes to state must survive crashes. Once a transaction is committed, the only way to change the state it produces is with another transaction.

In our web update example, the system must not return from the `commitTransaction()` call until all of the transaction’s updates have been safely stored in persistent storage.

Transactions v. critical sections The ACID properties are closely related to the properties of critical sections. Critical sections provide a way to update state that is atomic, consistent, and isolated but not durable. Adding the durability requirement significantly changes how we implement atomic updates.

Battling terminology

In operating systems, we use the term *consistency* in two ways. In the context of critical sections and transactions, we use “consistency” to refer to the idea of a system’s invariants being maintained (e.g., “are my data structures consistent?”) In the context of distributed memory machines and distributed systems, we use “consistency” to refer to the memory model—the order in which updates can become visible to reads (e.g., “are my system’s reads at different caches sequentially consistent?”).

Where there is potential confusion, we will use the terms *transaction consistency* or *memory model consistency*.

14.1.3 Implementing transactions

The challenge with implementing transaction is that we want a group of related writes to be atomic, but for persistent storage hardware like disks and flash, the atomic operation is a single-sector or single-page write. So, we must devise a way for a group of related writes to take effect when a single-sector write occurs.

If a system simply starts updating data structures in place, then it is vulnerable to a crash in the middle of a set of updates: the system has neither the complete set of old items (to roll back) nor a complete set of new items (to commit), so an untimely crash can force the system to violate atomicity.

Instead, a transactional system can persistently store all of a transaction’s **intentions**, the updates that will be made if the transaction commits, in some separate location of persistent storage. Only when all intentions are stored and the transaction commits should the system begin overwriting the target data structures; if the overwrites are interrupted in the middle, then on recovery the system can complete the transaction’s updates using the persistently stored intentions.

Redo logging

A common and very general way to implement transactions is redo logging. *Redo*

redo logging uses a persistent log for recording intentions and executes a transaction in three stages:

1. **Prepare.** Append all planned updates to the log.

This step can happen all at once, when the transaction begins to commit, or it can happen over time, appending new updates to the log as the transaction executes. What is essential is that all updates are safely stored in the log before proceeding to the next step.

2. **Commit.** Append a commit record to the log, indicating that the transaction has committed.

Of course a transaction may roll back rather than commit. In this case, a roll-back record may be placed in the log to indicate that the transaction was abandoned. Writing a roll-back record is optional, however, because a transaction will only be regarded as committed if a commit record appears in the log.

3. **Write-back.** Once the commit record is persistent in the log, all of a transaction's updates may be written to their target locations, replacing old values with new ones.

Once a transaction's write-back completes, its records in the log may be garbage collected.

The moment in step 2 when the sector containing the commit record is successfully stored is the *atomic commit*: before that moment the transaction may safely be rolled back; after that moment, the transaction must take effect.

Definition: atomic commit

Recovery. If a system crashes in the middle of a transaction, it must execute a recovery routine before processing new requests. For redo logging, the recovery routine is simple: scan sequentially through the log, taking the following actions for each type of record:

1. **Update record for a transaction.** Add this record to a list of updates planned for the specified transaction.
2. **Commit record for a transaction.** Write-back all of the transaction's logged updates to their target locations.
3. **Roll-back record for a transaction.** Discard the list of updates planned for the specified transaction.

When the end of the log is reached, the recovery process discards any update records for transactions that do not have commit records in the log.

Example. Consider, for example, a transaction that transfers \$100 from Tom's account to Mike's account. Initially, as Figure 14.2-(a) shows, data stored on disk and in the volatile memory cache indicates that Tom's account has \$200 and Mike's account has \$100.

Then, the cached values are updated and the updates are appended to the nonvolatile log (b). At this point, if the system were to crash, the updates in cache would be lost, the updates for the uncommitted transaction in the log would be discarded, and the system would return to its original state.

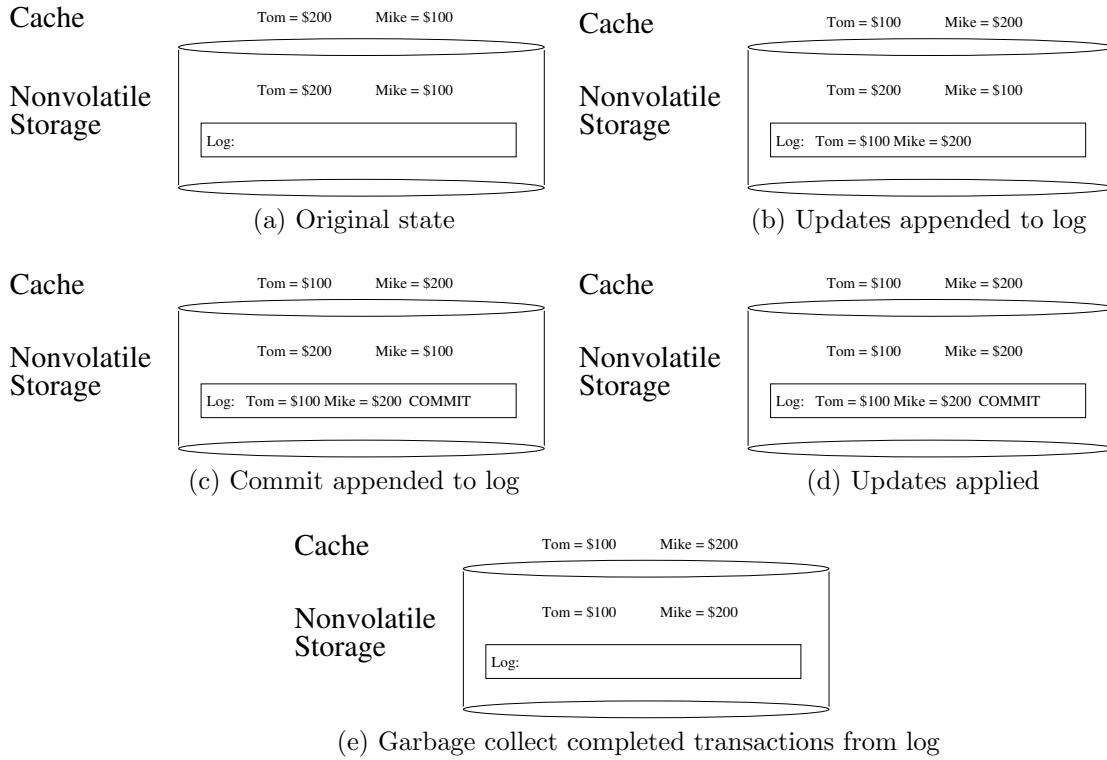


Figure 14.2: Example transaction with redo logging.

Once the updates are safely in the log, the commit record is appended to the log (c). This commit record should be written atomically based on the properties of the underlying hardware (e.g., by making sure it fits on a single disk sector and putting a strong checksum on it). This step is the atomic commit: prior to the successful storage of the commit record, a crash would cause the transaction to roll back; the instant the commit record is persistently stored, the transaction has committed and is guaranteed to be visible to all reads in the future. Even if a crash occurs, the recovery process will see the committed transaction in the log and apply the updates.

Now, the records in persistent storage for Tom and Mike's accounts can be updated (d).

Finally, once Tom and Mike's accounts are updated, the transaction's records in the log may be garbage collected (e).

Implementation details. A few specific techniques and observations are important for providing good performance and reliability for transactions with

redo logs.

- **Logging concurrent transactions.** Although the previous example shows a single transaction, multiple transactions may be executing at once. In these cases, each record in the log must identify the transaction to which it belongs.
- **Asynchronous write-back.** Step 3 of a transaction (*write-back*) can be asynchronous—once the updates and commit are in the log, the writeback can be delayed until it is convenient or efficient to perform it.

This flexibility yields two advantages. First, the latency from when a transaction calls `commit()` to when the call returns is minimized: as soon as the commit is appended to the sequential log, the call can return. Second, the throughput for writeback can be improved because the disk scheduler gets to operate on large batches of updates.

Two things limit the maximum write-back delay, but both are relatively loose constraints. First, larger write-back delays mean that crash recovery may take longer because there may be more updates to read and apply from the log. Second, the log takes space in persistent storage, which may in some cases be constrained.

- **Repeated write-backs are OK.** Some of the updates written back during recovery may already have been written back before the crash occurred. For example, in Figure 14.3 all of the records from the persistent log-head pointer to the volatile one have already been written back, and some of the records between the volatile log-head pointer may have been written back.

It is OK to reapply an update from a redo log multiple times because these updates are (and must be) *idempotent*—they have the same effect whether executed once or multiple times. For example, if a log record says “write ‘hello’ to the start of sector 74” then it doesn’t matter whether that value is written once, twice, or a hundred times to sector 74.

Definition: **idempotent**

Conversely, redo log systems cannot permit non-idempotent records such as “add 42 to each byte in sector 74.”

- **Restarting recovery is OK.** What happens if another crash occurs during recovery? When the system restarts, it simply begins recovery again. The same sequence of updates to committed transactions will be discovered in the log, and the same write-backs will be issued. Some of the write-backs may already have finished before the first crash or during some previous, but repeating them causes no problems.
- **Garbage collection constraints.** Once write-back completes and is persistently stored for a committed transaction, its space in the log can be reclaimed.

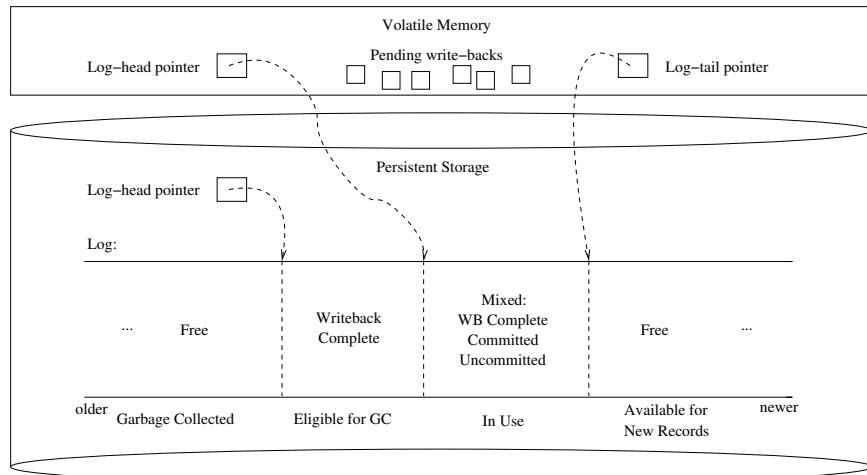


Figure 14.3: Volatile and persistent data structures for a transactional system based on a replay log.

For concreteness, Figure 14.3 illustrates a transaction log with an area of the log that is in use, an area that is no longer needed because it contains only records for transactions whose writebacks have completed, and an area that is free.

In this example system, the system's volatile memory maintains pointers to the head and tail of the log, new transaction records are appended to the tail of the log and cached in volatile memory, a write-back process asynchronously writes pending write-backs for committed transactions to their final locations in persistent storage, and a garbage collection process periodically advances a persistent log-head pointer so that recovery can skip at least some of the transactions whose writebacks are complete.

Example: New writes v. garbage collection.

Question: Suppose we have a circular log organized like the one in Figure 14.3. We must ensure that new records do not overwrite records that we may read during recovery, so we must ensure that the log-tail does not catch the log-head. But there are two log-heads, one in volatile memory and another in persistent storage. Which log-head represents the barrier that the log-tail must not cross?

Answer: The log-tail must not catch the persistent log-head pointer. Even though the records between the persistent and volatile log-heads have already been written back, during crash recovery, the recovery process will begin reading the log

Undo logging

Although transactions are often implemented with redo logging in which updates and the commit are written to the log and then the updates are copied to their final locations, transactions can also be implemented with *undo logging*.

To update an object, a transaction first writes the *old* version of the object to the log. It then writes the new version to its final storage location. When the transaction completes, it simply appends *commit* to the log. Conversely, if the transaction rolls back, the updates are undone by writing the old object versions to their storage locations.

The recovery process takes no action for committed transactions it finds in the log, but it undoes uncommitted transactions by rewriting the original object versions stored in the log.

Undo logging allows writes to objects to be sent to disk when they are generated and requires them to be stored on disk before a transaction is committed. This pattern is similar to update-in-place approaches, so in some cases it may be easier to add undo logging than redo logging to legacy systems. On the other hand, for storage systems like disks whose sequential bandwidth dominates their random I/O performance, undo logging may require more random I/Os before a transaction is committed (hurting latency) and it may give up chances to improve disk-head scheduling by writing large numbers of transactions' updates as a batch.

Undo/redo logging stores both the old and new versions of an object in the log. This allows updated objects to be written to their final storage locations whenever convenient, whether before or after the transaction is committed. If the transaction rolls back, any modified objects can be restored to the proper state, and if the system crashes, any committed transactions can have their updates redone and any uncommitted transactions can have their updates undone.

from the location indicated by the persistent log-head pointer.
As long as the records are intact, recovering from the persistent log-head pointer rather than the volatile one may waste some work, but it will not affect correctness.

- **Ordering is essential.** It is vital to make sure that all of a transaction's updates are on disk in the log before the commit is, that the commit is on disk before any of the write-backs are, and that all of the write-backs are before a transaction's log records are garbage collected.

In Linux, an application can call `sync()` or `fsync()` to tell the operating system to force buffered writes to disk. These calls return only once the updated blocks are safely stored. Within the operating system, a request can be tagged with a `BIO_RW_BARRIER` tag, which ensures that all preceding writes and no subsequent ones are stored to disk before the tagged request is.

Isolation and concurrency revisited. Redo logging provides a mechanism for atomically making multiple updates durable, but if there are concurrent transactions operating on shared state, we must also ensure isolation—each transaction must appear to operate on its own.

Definition: **two-phase locking**

A common way to enforce isolation among transactions is *two-phase locking*, which divides a transaction into two phases. During the *expanding* phase, locks may be acquired but not released. Then, in the *contracting* phase, locks may be released but not acquired. In the case of transactions, because we want isolation and durability, the second phase must wait until after the transaction commits or rolls back so that no other transaction sees updates that later disappear.

Definition: **serializability**

Serializability across transactions ensures that the result of any execution of the program is equivalent to an execution in which transactions are processed one at a time in some sequential order. So, even if multiple transactions are executed concurrently, they can only produce results that they could have produced had they been executed one at a time in some order.

Although acquiring multiple locks in arbitrary orders normally risks deadlock, transactions provide a simple solution. If a set of transactions deadlocks, one or more of the transactions can be forced to roll back, release their locks, and restart at some later time.

Performance of redo logging. It might sound like redo logging will impose a significant performance penalty compared to simply updating data in place: redo logging writes each update twice—first to the log and then to its final storage location.

Things are not as bad as they initially seem. Redo logging can have excellent performance—often better than update in place—especially for small writes. Four factors allow efficient implementations of redo logging:

- **Log updates are sequential.** Because log updates are sequential, appending to the log is fast. With spinning disks, large numbers of updates can be written as a sequential stream without seeks or rotational delay once the write begins. Many high-performance systems dedicate a separate disk for logging so that log appends never require a seek. For flash storage, sequential updates are often significantly faster than random updates, though the advantage is not as pronounced.
- **Writeback is asynchronous.** Because writeback can be delayed until some time after a transaction has been committed, transactions using redo logs can have good response time (because the transaction commit only requires appending a commit record to the log) and can have good throughput (because batched writebacks can be scheduled more efficiently than individual or small groups of writes that must occur immediately.)

Multiversion Concurrency Control

An alternative to enforcing transaction isolation with locks is to enforce it with multiversion concurrency control. In *multiversion concurrency control* each write of an object x creates new version of x , the system keeps multiple versions of x , and the system directs each read to a specific version of x . By keeping multiple versions of objects, the system can allow transaction A to read a version of x that has been overwritten by transaction B even if B needs to be serialized after A .

There are various multiversion concurrency control algorithms that ensure serializability. A simple one is multiversion timestamp ordering (MVTO), which processes concurrent transactions, enforces serializability, never blocks a transaction's reads or writes, but which may cause a transaction to roll back if it detects that a read of a later transaction (based on the serializable schedule MVTO is enforcing) was executed before—and therefore did not observe—the write of an earlier transaction (in serialization order.)

MVTO assigns each transaction T a logical timestamp. Then, when T writes an object x , MVTO creates a new version of x labeled with T 's timestamp t_T , and when T reads an object y , MVTO returns the version of y , y_v with the highest timestamp that is at most T 's timestamp; MVTO also makes note that y_v was read by transaction t_T . Finally, when T attempts to commit, MVTO blocks the commit until all transactions with smaller timestamps have committed or aborted.

MVTO rolls back transaction rather than allowing it to commit in three situations. First, If MVTO aborts any transaction, it removes the object versions written by that transaction and rolls back any transactions that read those versions. Notice that a transaction that reads a version must have a higher timestamp than the one that wrote it, so no committed transactions need to be rolled back.

Second, if a transaction T writes an object that has already been read by a later transaction T' that observed the version immediately prior to T 's write, T MVTO rolls back T . It does this because if T were to commit, T' 's read must return T 's write, but that did not occur.

Third, if MVTO garbage collects old versions and transaction T reads an object for which the last write by an earlier transaction has been garbage collected, then MVTO rolls back T .

Additional details and alternative implementations of multiversion concurrency control for transactions are available in *Concurrency Control and Recovery in Database Systems* by Philip A. Bernstein, Vassos Hadzilacos, and Nathan Goodman (chapter 5).

Relaxing Isolation

In this book we focus on the strong and relatively simple isolation requirement of serializability: no matter how much concurrency there is, the system must ensure that the results of any execution of the program is equivalent to an execution in which transactions are processed one at a time in some sequence. However, strong isolation requirements sometimes force transactions to block (e.g., when waiting to acquire locks) or roll back (e.g., when fixing a deadlock or encountering a “late write” under multiversion concurrency control).

Relaxing the isolation requirement can allow effectively higher levels of concurrency by reducing the number of cases in which transactions must block or roll back. The cost, of course, is potentially increased complexity in reasoning about concurrent programs, but several relaxed isolation semantics have proven to be sufficiently strong to be widely used.

For example, *snapshot isolation* requires each transaction’s reads appear to come from a snapshot of the system’s committed data taken when the transaction starts. Each transaction is buffered until the transaction commits, at which point the system checks all of the transaction’s updates for *write-write conflicts*. A write-write conflict occurs if transaction T reads an object o from a snapshot at time t_{start} and tries to commit at time t_{commit} but some other transaction T' commits an update to o between T ’s read at t_{start} and T ’s attempted commit at t_{commit} . If a write-write conflict is detected for any object being committed by T , T is rolled back.

Snapshot isolation is weaker than serializability because each transaction’s reads logically happen at one time and its writes logically happen at another time. This split allows, for example, *write skew anomalies* where one transaction reads object x and updates object y and a concurrent transaction reads object y and updates object x . If there is some constraint between x and y , it may now be violated. For example, if x and y represent the number of hours two managers have assigned you to work on each of two tasks with a constraint that $x + y \leq 40$. Manager 1 could read $x = 15$ and $y = 15$, attempt to assign 10 more hours of work on task x , and verify that $x + y = 25 + 15 \leq 40$. In the mean time manager 2 could read $x = 15$ and $y = 15$, attempt to assign 10 more hours of work on task y , verify that that $x + y = 15 + 25 \leq 40$, and successfully commit the update, setting $y = 25$. Finally, manager 1 could successfully commit its update, setting $x = 25$ and ruining your weekend.

- **Fewer barriers or synchronous writes are required.** Some systems avoid using transactions by carefully ordering updates to data structures so that they can ensure that if a crash occurs, a recovery process will be able to scan, identify, and repair inconsistent data structures. However, these techniques often require large number of barrier or synchronous write operations, which reduce opportunities to pipeline or efficiently schedule updates.

In contrast, transactions need a relatively small number of barriers: one after the updates are logged and before the commit is logged, another after the commit is logged but before the transaction is reported as successful (and before writebacks begin), and one after a transaction's writebacks complete but before the transaction's log entries are garbage collected.

- **Group commit.** *Group commit* is often used to further improve transactions' performance. Group commit techniques combine a set of transactions' commits into one log write to amortize the cost of initiating the write (e.g., seek and rotational delays). Group commit techniques can also be used to reduce the number of barrier or sync operations needed to perform a group of transactions.

Definition: **group commit**

Example: Performance of small-write transactions.

Suppose you have a 1TB disk that rotates once every 10 ms, that has a maximum sustained platter transfer rate of 50 MB/s for inner tracks and 100 MB/s for outer tracks, and that has a 5 ms average seek time, a 0.5 ms minimum seek time, and a 10 ms maximum seek time.,.

Consider updating 100 randomly selected 512-byte sectors in place with the total time to first commit the updates to a log; assume that the updates must be ordered for safety (e.g., update i must be on disk before update $i + 1$ is.)

Question: Compare the total time to complete these updates with a simple update in place approach with the cost when using transactions implemented with a redo log.

Answer: Using a simple update in place approach, we need to use FIFO scheduling to ensure updates hit the disk in order, so the time for each update is approximately $\text{average seek time} + 0.5 \text{ rotation time} + \text{transfer time} = 5 \text{ ms} + 5 \text{ ms} + \text{transfer time}$. Transfer time will be at most $\frac{512}{50*10^6}$ seconds, which will be negligible compared to the other terms. So, we have $10 \text{ ms per request}$ or $1 \text{ s for 100 requests}$ for update in place.

For transactions, we first append the 100 writes to the log. We will conservatively assume that each update consumes 2 sectors (one for the data and the other for metadata indicating the transaction number, the target sector on disk, etc.) So, assuming that the disk head is at a random location when the request arrives,

our time to log the requests is $\text{average seek time} + 0.5 \text{ rotation time} + \text{transfer time} = 5 \text{ ms} + 5 \text{ ms} + \frac{200 \times 512}{100 \times 10^6} = 10.24 \text{ ms}$.

Next, we need to append the commit record to the transaction. If the disk hardware supports a barrier instruction to enforce ordering of multiple in-progress requests, the operating system can issue this request along with the 100 writes. Here, we'll be conservative and assume that the system does not issue the commit's write until after the 100 writes in the body of the transaction are in the log. So, we will likely have to wait one full revolution of the disk to finish the commit: 10 ms .

Finally, we need to write the 100 writes to their target locations on disk. Unlike the case for update in place, ordering does not matter here, so we can schedule them and write them more efficiently. Estimating this time takes engineering judgement, and different people are likely to make different estimates. For this example, we will assume that the disk uses a variant of shortest service time first (SSTF) scheduling in which the scheduler looks at the four requests on the next nearest tracks and picks the one with the shortest predicted seek time + rotational latency from the disk head's current position. Because the scheduler gets to choose from four requests, we will estimate that the average rotational latency will be one forth of a rotation, 2.5 ms ; this may be conservative since it ignores the fact that request i will always remove from the four requests being considered the one that would have been rotationally farthest away if it were an option for request $i + 1$. Because we initially have 100 requests and because we are considering the four requests on the nearest tracks, the farthest seek should be around 4% of the way across the disk, and the average one to a member of the group being considered should be around 2%. We will estimate that seeking 2-4% of the way across disk costs twice the minimum seek time: 1 ms .

Putting these estimates for writeback time together, the writebacks of the 100 sectors should take about $\text{estimated scheduled seek time} + \text{estimated scheduled rotational latency} = 1.0 \text{ ms} + 2.5 \text{ ms} = 3.5 \text{ ms}$ per request or 350 ms total.

Adding the logging, commit, and writeback times, we have $10.24 \text{ ms} + 10 \text{ ms} + 350 \text{ ms} = 370.24 \text{ ms}$. The transactional approach is almost three times faster even though it writes the data twice and even though it provides the stronger atomic-update semantics.

Question: For the same two approaches, compare the response time latency from when a call issuing these requests is issued until that call can safely return because all of the updates are durable.

Answer: The time for update in place is the same as above: 1 s . The time

for the transactional approach is the time for the first two steps: logging the updates and then logging the commit: $10.24\text{ ms} + 10\text{ ms} = 20.24\text{ ms}$.

Although small writes using redo logging may actually see performance benefits compared to update in place approaches, large writes may see significant penalties.

Example: Performance of large-write transactions.

Question: Considering the same disk and approaches as in the example above, compare the total time to for 100 writes, but now assume that each of the 100 writes updates a randomly selected 1 MB range of sequential sectors.

Answer: For the update in place approach, the time for each update is approximately $\text{average seek time} + 0.5 \text{ rotation time} + \text{transfer time} = 5\text{ ms} + 5\text{ ms} + \text{transfer time}$. We will assume that the bandwidth for an average transfer is 75 MB/s—between the 50 MB/s and 100 MB/s inner and out tracks' transfer rates. So, we estimate the average transfer time to be $100\text{ MB}/75\text{ MB/s} = 1.333\text{ s}$, giving a total time of $.005\text{ s} + .005\text{ s} + 1.333\text{ s} = 1.343\text{ s}$ per request and 134.3 s for 100 requests.

For the transactional approach, our time will be $\text{time to log updates} + \text{time to commit} + \text{time to write back}$.

For logging the updates, we'll assume a reasonably efficient encoding of metatadata that makes the size of the metadata for a 100 MB sequential update negligible compared to the data. So, logging the data will take $\text{seek time} + \text{rotational latency} + \text{transfer time} = 5\text{ ms} + 5\text{ ms} + 100 * 100\text{ MB}/100\text{ MB/s} = .005\text{ s} + .005\text{ s} + 100\text{ s} \approx 100\text{ s}$.

Writing the commit adds another 10 ms as in the above example.

Finally, as above, doing the write-backs $\text{estimated scheduled seek time} + \text{estimated scheduled latency} + \text{transfer time} = 1.0\text{ ms} + 2.5\text{ ms} + 100\text{ MB}/75\text{ MB/s}$, giving a total of 1.337 s per request and 133.7 s for 100 requests.

Adding the data logging, commit, and writeback times together, the transactional approach takes about 233 seconds while the update in place approach takes about 134 seconds. In this case transactions do impose a significant cost, nearly doubling the total time to process these updates.

Question: Now compare the latency from when the call making the 100 writes is issued until it may safely return.

Answer: Under the update in place approach, we can only return when everything is written, while under the transactional approach, we can return once the commit is complete. So, we have comparable times: 134 s for update in place and 100 s for transactions.

One way to reduce transaction overheads for large writes is to add a level of indirection: write the sequential data to a free area of the disk, but not in the circular log where it would have to be reclaimed later. Then, the update in the log just needs to be a reference to that data rather than the data itself. Finally, after the transaction commits, perform the writeback by updating a pointer in the original data structure to point to the new data.

14.1.4 Transactions and file systems

transactions + file systems

[[motivation]]

logging v. journaling v. update-in place

write-anywhere file systems revisited

Application-level transactions

application use of transactions
-- log + replay
-- intentions, update
--

cautionary tail -- "sync" buried in library

the one place you do see it is application level -- save new file,
rename
e.g., emacs looking for #emacs# file

"intentions to avoid scanning is like writeahead log"

point out that to make this go fast, need to avoid scanning everything
 --> write down what you're planning to modify in well known location
 so that on restart you can check to see what needs to check...
 this ends up being like a writeahead log

painful example: remzi's paper -- apple flush after each update

```
-- this is what many file systems used to do
-- as discussed in the sidebar, approach largely
abandoned for a number of reasons;
-- one problem: similar to "too much milk" -- complex and
error prone; want simpler, more principled approach
```

Explicit logging database example

word document example

Implicit logging Key idea: need to record intentions before changing state

[[we did atomic swap above, right?]]

Exercises

1. Suppose that a text editor application uses the `rename` technique discussed on page 403 for safely saving updates by saving the updated file to a new file (e.g., `#doc.txt#` and then calling `rename(''#doc.txt'', ''doc.txt'')` to change the name of the updated file from `#doc.txt#` to `doc.txt`. Posix `rename` promises that the update to `doc.txt` will be atomic—even if a crash occurs, `doc.txt` will refer to either the old file or the new one. However, Posix does not guarantee that the entire `rename` operation will be atomic. In particular, Posix allows implementations in which there is a window in which a crash could result in a state where both `doc.txt` and `#doc.txt#` refer to the same, new document.
 - a. How should a text-editing application react if, on startup, it sees both `doc.txt` and `doc.txt` and (i) both refer to the same file or (ii) each refers to a file with different contents?
 - b. Why might Posix permit this corner case (where we may end up with two names that refer to the same file) to exist?
 - c. Explain how an FFS-based file system without transactions could use the “ad hoc” approach discussed in Section 14.1.1 to ensure that (i) `doc.txt` always refers to either the old or new file, (ii) the new file is never lost – it is always available as at least one of `doc.txt` or `#doc.txt#`, and (iii) there is some window where the new file may be accessed as both `doc.txt` and `#doc.txt#`.

- d. Section 14.1.1 discusses three reasons that few modern file systems use the “ad-hoc” approach. However, many text editors still do something like this. Why have the three issues had less effect on applications like text editors than on file systems?
2. Above, we defined two-phase locking for basic mutual exclusion locks. Extend the definition of two-phase locking for systems that use readers-writers locks.
3. Suppose that x and y represent the number of hours two managers have assigned you to work on each of two tasks with a constraint that $x + y \leq 40$. On page 414, we showed that snapshot isolation could allow one transaction to update x and another concurrent transaction to update y in a way that would violate the constraint $x + y \leq 40$. Is such an anomaly possible under serializability? Why or why not?
4. Suppose you have transactional storage system `tStore` that allows you to read and write fixed-sized 2048-byte blocks of data within transactions, and you run the following code.

```

...
byte b1[2048];  byte b2[2048];
byte b3[2048];  byte b4[2048];

TransID t1 = tStore.beginTransaction();
TransID t2 = tStore.beginTransaction();
TransID t3 = tStore.beginTransaction();
TransID t4 = tStore.beginTransaction();

// Interface is
//     writeBlock(TransID tid, int blockNum, byte buffer[]);
tStore.writeBlock(t1, 1, ALL_ONES);
tStore.writeBlock(t1, 2, ALL_TWOS);
tStore.writeBlock(t2, 3, ALL_THREES);
tStore.writeBlock(t1, 3, ALL_FOURS);
tStore.writeBlock(t1, 2, ALL_FIVES);
tStore.writeBlock(t3, 2, ALL_SIXES);
tStore.writeBlock(t4, 4, ALL_SEVENS);
tStore.readBlock(t2, 1, b1);
tStore.commit(t3);
tStore.readBlock(t2, 3, b2);
tStore.commit(t2);
tStore.readBlock(t1, 3, b3);
tStore.readBlock(t4, 3, b4);
tStore.commit(t1);

// At this point, the system crashes

```

The system crashes at the point indicated above.

Assume that `ALL_ONES`, `ALL_TWOS`, etc. are each arrays of 2048 bytes with the indicated value. Assume that when the program is started, all blocks in the `tStore` have the value `ALL_ZEROS`.

Just before the system crashes, what is the value of `b1` and what is the value of `b2`?

- (a) In the program above, just before the system crashes, what is the value of `b3` and what is the value of `b4`?
- (b) Suppose that after the program above runs and crashes at the indicated point. After the system restarts and completes recovery and all write-backs, what are the values stored in each of blocks 1, 2, 3, 4, and 5 of the `tStore`?

PROBLEM:

```
[[ compare performance of 1000 updates in place v. transaction --
  fall 2011 exam ]]
```

14.2 Error detection and correction

Because data storage hardware is imperfect, storage systems must be designed to detect and correct errors. Storage systems take a layered approach:

- Storage hardware detects many failures with checksums and device-level checks, and it corrects small corruptions with error correcting codes
- Storage systems include redundancy using RAID architectures to reconstruct data lost by individual devices
- Many recent file systems include additional end-to-end correctness checks

These techniques are essential. Essentially all persistent storage devices include internal redundancy to achieve high storage densities with acceptable error rates, but the limits of this internal redundancy are significant enough that it is difficult to imagine designing a storage system for important data without additional redundancy for error correction, and it is hard to think of a significant file system developed in the last decade that does not include higher-level checksums.

Though essential and widespread, there are significant pitfalls in designing and using these techniques. In our discussions, we will point out issues that, if not handled carefully, can drastically reduce reliability.

The rest of this section examines error detection and correction for persistent storage, starting with the individual storage devices, then examining how RAID replication helps tolerate failures by individual storage devices, and finally looking at the end-to-end error detection in many recent file systems.

What causes sector or page failures?

For spinning disks, permanent sector failures can be caused by a range of faults such as pits in the magnetic coating where a contaminant flaked off the surface, scratches in the coating where a contaminant was dragged across the surface by the head, or smears of machine oil across some sectors of a disk surface.

Transient sector faults, where a sector's stored data is corrupted but where new data can be successfully written to and read from the sector, can be caused by factors such as write interference where writes to one track disturb bits stored on nearby tracks and "high fly writes" where the disk head gets too far from the surface, producing magnetic fields too weak to be accurately read.

For flash storage, permanent page failures can be caused by manufacturing defects or by wear-out when a page experience a large number of write/erase cycles.

Transient flash storage failures can be caused by write disturb errors where charging one bit also causes a nearby bit to be charged, read disturb errors where repeatedly reading one page changes values stored on a nearby page, over-programming errors where too high a voltage is used to write a cell, which may cause incorrect reads or writes, and data retention error where charge may leak out of or into a flash cell over time, changing its value; wear-out from repeated write/erase cycles can make devices more susceptible to data retention errors.

14.2.1 Storage device failures and mitigation

Storage hardware pushes the limits of physics, material sciences, and manufacturing processes to maximize storage capacity and performance. These aggressive designs leave little margin for error, so manufacturing defects, contamination, or wear can cause stored bits to be lost.

Individual spinning disks and flash storage devices exhibit two types of failure. First, isolated disk sectors or flash pages can lose existing data or degrade to the point where they cannot store new data. Second, an entire device can fail, preventing access to all of its sectors or pages. We discuss each of these in turn to understand the problems higher level techniques need to deal with.

Sector and page failures

Definition: **sector failure** Disk *sector failures* occur when data on one or more individual sectors of a disk are lost, but the rest of the disk continues to operate correctly. Flash *page failures* are the equivalent for flash pages.

Storage devices use two techniques to mitigate sector or page failures: error correcting codes and remapping.

Mitigation: Error correcting codes. *Error correcting codes* deal with failures when some of the bits in a sector or page are corrupted. When the device stores data, it encodes the data with additional redundancy. Then, if a small number of bits are corrupted in a sector or page being read, the hardware automatically corrects the error, and the read successfully completes. If the damage is more extensive, then with high likelihood the read fails and returns an error code; being told that the device has lost data is not a perfect solution, but it is better than having the device silently return the wrong data.

Definition: **error correcting codes**

Manufacturers balance storage space overheads against error correction capabilities to achieve acceptable advertised sector or page failure rate, typically expressed as the expected number of bits that can be read before encountering an unreadable sector or page. In 2011, advertised disk and flash *nonrecoverable read error* rates typically range between one sector or page per 10^{14} to 10^{16} bits read. The nonrecoverable read error rate is sometimes called the *bit error rate*.

Definition: **nonrecoverable read error**

Definition: **bit error rate**

Mitigation: Remapping Disks and flash are manufactured with some number of spare sectors or pages so that they can continue to function despite some number of permanent sector or page failures by remapping failed sectors or pages to good ones. Before shipping hardware to users, manufacturers scan devices to remap bad sectors or pages caused by manufacturing defects. Later, if additional permanent failures are detected, the operating system or device firmware can remap the failed sectors or pages to good ones.

Pitfalls. Although devices' nonrecoverable read rate specifications are helpful, designers must avoid a number of common pitfalls:

- **Assuming that nonrecoverable read error rates are negligible.** Storage devices' advertised error rates sound impressive, but with the large capacities of today's storage, these error rates are non-negligible. For example, if you completely read a 2 TB disk with a bit error rate of 1 sector per 10^{14} bits, there may be more than a 10% chance of encountering at least one error.
- **Assuming nonrecoverable read error rates are constant.** Although a device may specify a single number as its unrecoverable read error rate, many factors can affect the rate at which such errors manifest. A given device's actual bit error rate may depend on its load (e.g., some faults may be caused by device activity), its age (e.g., some faults may become more likely as a device ages), or even its specific workload (e.g., faults in some sectors or pages may be caused by reads or writes to nearby sectors or pages.)
- **Assuming independent failures.** Errors may be correlated in time or space: finding an error in one sector may make it more likely that you

will find one in a nearby sector or that you will to find a fault in another sector soon.

- **Assuming uniform error rates.** The relative contributions of different causes of nonrecoverable read errors can vary across models and different generations or production runs of the same model. For example, one model of disk drive might have many of its sector read errors caused by contaminants damaging its recording surfaces while another model might have most of its errors caused by write interference where writes to one track perturb data stored on nearby tracks. The first might see its error rate rise over time, while the second might have an error rate that increases as its write/read ratio increases.

Failure rates can even vary across different individual devices. If you deploy several outwardly identical disks, some may exhibit tens of nonrecoverable read errors in a year, while others operate flawlessly.

Example: Unrecoverable read errors.

Question: Suppose that the nearly-full 500 GB disk on your laptop has just stopped working. Fortunately, you have a recent, full backup on a 500 GB USB drive with an unrecoverable read error rate of 1 sector per 10^{14} bits read. Estimate the probability of successfully reading the entire USB backup disk when restoring your data to a replacement laptop disk.

Answer: We need to read 500 GB, so the expected number of failures is $500 \text{ GB} * \frac{8*10^9 \text{ bits}}{\text{GB}} * \frac{1 \text{ error}}{10^{14} \text{ bits}} = 0.04$. The probability of encountering at least one failure might be a bit lower than that (since we may encounter multiple failures as we scan the entire disk), but there appears to be a chance of at least a few percent that the restoration will not be fully successful.

We can approach the problem in a slightly different way by interpreting the unrecoverable read rate as meaning that each bit has a 10^{-14} chance of being wrong and that failures are independent (both somewhat dubious assumptions, but probably OK for a ballpark estimate). Then each bit has a $1 - 10^{-14}$ chance of being correct, and the chance of reading all bits successfully is $P_S = (1 - 10^{-14})^{8*500*10^9} = 0.9608$. Under this calculation, we estimate that there is slightly less than a 4% chance of encountering a failure during the full-disk read of the backup disk.

As noted in the sidebar, these calculations ignore some important factors, so the results may not be precise. But, even if they are off by as much as an order of magnitude, then it is still reasonable to conclude that the rate of nonrecoverable read errors is likely to be non-negligible.

What causes whole-device failures?

Disk failures can be caused by a range of faults such as a disk head being damaged, a capacitor failure or power surge that damages the electronics, or mechanical wear-out that makes it difficult for the head to stay centered over a track.

Common causes of flash device failures include wear-out, when enough individual pages fail that the device runs out of spare pages to use for remapping, and failures of the device's electronics such as having a capacitor fail.

Note that the impact of a small number of lost sectors may be modest (e.g., the backup software succeeds in restoring all but a file or two) or it may be severe (e.g., no data is restored.) For example, if the sector failure corrupts the root directory, a significant fraction of the data may be lost.

Device failures

Full disk or flash drive failures are when a device stops being able to service reads or writes to all sectors.

Definition: **disk device failure, flash device failure**

When a whole device fails, the host computer's device driver will detect the failure, and reads and writes to the device will return error codes rather than, for example, returning incorrect data. This explicit failure notification is important because it reduces the amount of cross-device redundancy needed to correct failures.

Full device failure rates are typically characterized by an *annual failure rate*, the fraction of disks expected to fail each year, or by a *mean time before failure (MTTF)* which is the inverse of the specified constant annual failure rate. In 2011, specified annual failure rates (or MTTFs) for spinning disks typically range from 0.5% ($1.7 * 10^6$ hours) to 0.9% ($1 * 10^6$ hours); specified failure rates for flash solid state drives are similar.

Definition: **annual failure rate**

Definition: **mean time before failure (MTTF)**

Systems with many storage devices expect to encounter frequent failures. For example,

Pitfalls. Storage system designers must consider several pitfalls when considering advertised device failure rates.

- **Relying on advertised failure rates.** Studies across several large collections of spinning disks have found significantly variability in failure rates. In these studies, many systems experienced failure rates of 2%, 4%, or higher despite advertised failure rates of under 1%.

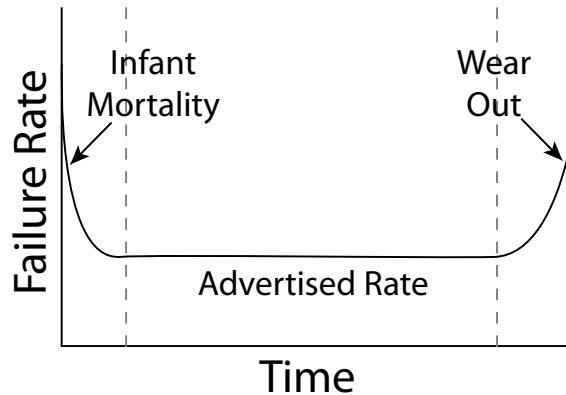


Figure 14.4: Bathtub model of device lifetimes

Some of the discrepancy may be due to different definitions of “failure” by manufacturers and users, some may be due to challenging field conditions, and some may be due to the limitations of the accelerated-aging and predictive techniques used by manufacturers to estimate MTTF.

- **Assuming uncorrelated failures.** Evidence from deployed systems suggests that when one fault occurs, other nearby devices are more likely to fail soon. Many factors can cause such correlation. For example, manufacturing irregularities can cause a batch of disks to be substandard, and an organization that purchases many disks from the same vendor at the same time may find themselves installing a batch of disks likely to fail in similar ways. As another example, disks in the same machine or rack may be of a similar age, may experience similar environmental stress and workloads, and may wear out at a similar time.
 - **Confusing a device’s MTTF with its useful life.** If a device has an MTTF of one million or more hours, it does not mean that it is expected to last for 100 years or more. Disks are designed to be operated for some finite lifetime, perhaps 5 years. A disk’s advertised annual failure rate (i.e., $1/\text{MTTF}$) applies during the disk’s intended service life. As that lifetime is approached, failure rates may rise as the device wears out.
 - **Assuming constant failure rates.** A device may have different failure rates over its lifetime. Some devices exhibit *infant mortality*, where their failure rate may be higher than normal during their first few weeks of use as latent manufacturing defects are exposed. Others exhibit *wear out*, where their failure rate begins to rise after some years.
- A simple model for understanding infant mortality and wear out is the *bathtub model* illustrated in Figure 14.4.
- **Ignoring warning signs.** Some device failures happen without warning,

Definition: **infant mortality**

Definition: **wear out**

Definition: **bathtub model**

but others are preceded by increasing rates of non-fatal anomalies. Many storage devices implement the *SMART (Self-Monitoring, Analysis, and Reporting Technology)* interface, which provides a way for the operating system to monitor events that may be useful in predicting failures such as read errors, sector remappings, inaccurate seek attempts, or failures to spin up to the target speed.

Definition: SMART
(Self-Monitoring,
Analysis, and Reporting
Technology)

- **Assuming devices behave identically.** Different device models or even different generations of the same model may have significantly different failure behaviors. One generation might exhibit significantly higher failure rates than expected and the next might exhibit significantly lower rates.

Example: Disk failures in large systems.

Question: Suppose you have a departmental file server with 100 disks, each with an estimated MTTF of $1.5 * 10^6$ hours. Estimate the expected time until one of those disks fails. For simplicity, assume that each disk has a constant failure rate and that disks fail independently.

Answer: If each disk has a MTTF of $1.5 * 10^6$ hours, then 100 disks fail at a 100 times greater rate, giving us a MTTF of $1.5 * 10^4$ hours. So, although the annual failure rate of a single disk is $\frac{1\text{failure}}{1.5*10^6\text{hours}} * \frac{24\text{ hours}}{\text{day}} * \frac{365\text{ days}}{\text{year}} = 0.00585\frac{\text{failures}}{\text{year}}$, the annual failure rate of the 100-disk system is 0.585 = 58.5%.

Example: Pitfalls.

Question: Given the pitfalls discussed above, is this calculation above likely to overestimate or underestimate the failure rate of the system?

Answer: Of the factors listed above, the pitfall of *relying on advertised failure rates* seems most significant, and it could lead us to significantly *underestimate* the failure rate of the system.

This solution does *assume constant failure rates*. If the disks are very new or very old, they may suffer higher failure rates than expected, which might cause us to *underestimate* the failure rate of the system.

Because we are only interested in the average rate, the *correlation* pitfall is not particularly relevant to our analysis.

The exponential distribution

When—as in the example—device failures events occur at a constant rate, the number of failure events in a fixed time period can be mathematically modelled as a Poisson process, and the interarrival time between failure events follows an exponential distribution.

The exponential distribution is *memoryless*—since the rate of failure events is constant across time, then the expected time to the next failure event is the same—no matter what the current time and no matter how long it has been since the last failure. So, if a device has an annual failure rate of 0.5 and thus a mean time to failure of 2 years, and we've been operating the device without a failure for a year, the expected time from the current time to the next failure is still 2 years.

So, if random variable T represents the time between failures and has an exponential distribution with λ representing the average number of failure events per unit of time, then the probability density function $f_T(t)$ is:

$$f_T(t) = \begin{cases} \lambda e^{-\lambda t} & \text{if } t \geq 0 \\ 0 & \text{if } t < 0 \end{cases}$$

and the mean time to failure is $MTTF = \frac{1}{\lambda}$.

Exponential distributions have a number of convenient mathematical properties. For example, because the failure rate is constant, the mean time to failure is the inverse of the failure rate; this is why it is easy to convert between MTTF and annual failure rates in storage specifications. Also, if the expected number of failures is given for one duration (e.g., 0.1 failures per year), it can easily be converted to the expected number for a different duration (e.g., 0.0003 failures per day). Finally, if we have k independent failure processes with rates of $\lambda_1, \lambda_2, \dots, \lambda_k$, then the aggregate failure function—the rate at which failures of any of the k kinds occurs—is

$$\lambda_{tot} = \lambda_1 + \lambda_2 + \dots + \lambda_k$$

and the mean time to the next failure of any kind is $MTTF_{tot} = \frac{1}{\lambda_{tot}}$. For example, if we have 100 disks, each with a $MTTF_{disk} = 1.5 * 10^6$ hours or, equivalently, each failing at a rate of 0.00585 failures per year, then the overall 100-disk system suffers failures at a rate of $100 * 0.00585 = 0.585$ failures per year or, equivalently, the 100-disk system has $MTTF_{100disks} = 1.5 * 10^4$ hours.

Warning. Because the exponential distribution is so mathematically convenient, is tempting to use it even when it is not appropriate. Remember that failures in real systems may be correlated (i.e., they are not independent) and may vary over time (i.e., they are not constant).

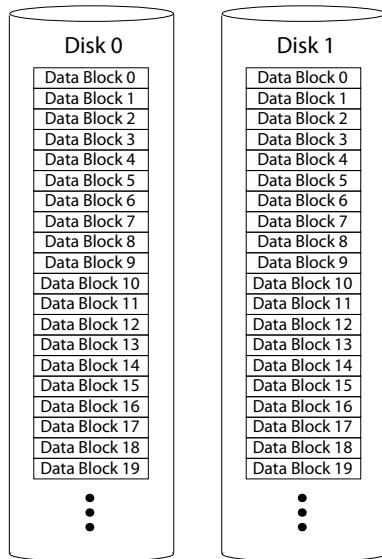


Figure 14.5: RAID 1 with mirroring.

14.2.2 RAID: Multi-disk redundancy for error correction

Given the limits of physical storage devices, storage systems use additional techniques to get acceptable end-to-end reliability. In particular, rather than trying to engineer perfectly reliable (and extremely expensive) storage devices, storage systems use Redundant Arrays of Inexpensive Disks (RAIDs) so that a partial or total failure of one device will not cause data to be lost.

Basic RAIDs

A *Redundant Array of Inexpensive Disks (RAID)* is a system that spreads data redundantly across multiple disks in order to tolerate individual disk failures. Note that the term RAID traditionally refers to redundant *disks*, and for simplicity we will discuss RAID in the context of disks. The principles, however, apply equally well to other storage devices like flash drives.

Definition: **RAID**

Figures 14.5 and 14.6 illustrate two common RAID architectures: mirroring and rotating parity.

Definition: **mirroring**

Definition: **RAID 1**

- **Mirroring.** In RAIDs with *mirroring* (also called *RAID 1*), the system writes each block of data to two disks and can read any block of data from either disk as Figure 14.5 illustrates. If one of the disks suffers a sector or whole-disk failure, no data is lost because the data can still be read from the other disk.

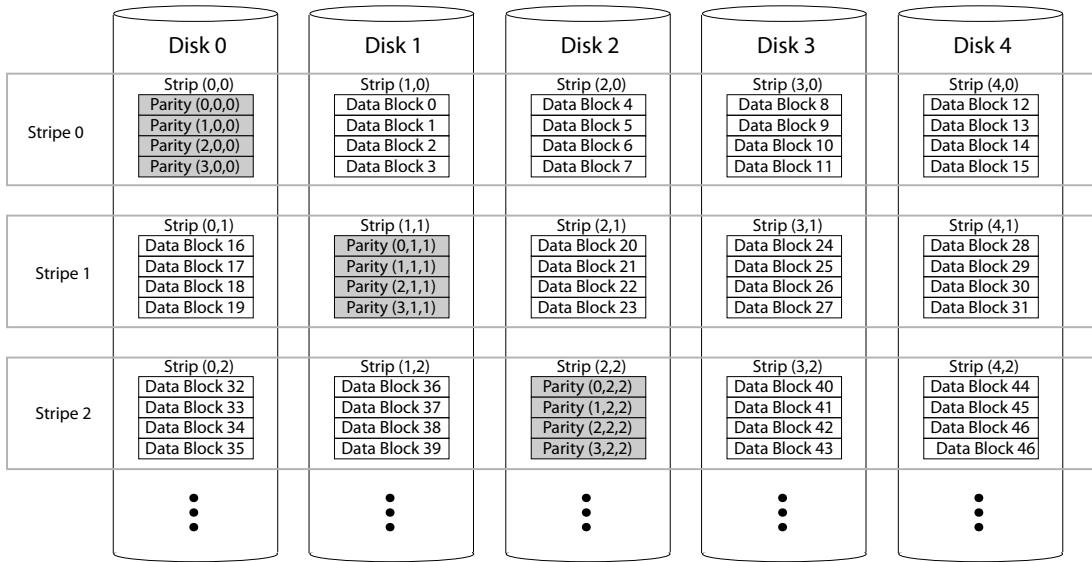


Figure 14.6: RAID 5 with rotating parity.

Definition: **rotating parity**

Definition: **RAID 5**

- **Rotating parity.** In RAIDs with *rotating parity* (also called *RAID 5*), the system reduces replication overheads by storing several blocks of data on several disks and protecting those blocks with one redundant block stored on yet another disk as Figure 14.6 illustrates.

In particular, this approach uses groups of G disks, and writes each of $G - 1$ blocks of data to a different disk and 1 block of parity to the remaining disk. Each bit of the parity block is produced by computing the exclusive-or of the corresponding bits of the data blocks:

$$\text{parity} = \text{data}_0 \oplus \text{data}_1 \oplus \dots \oplus \text{data}_{G-1}$$

If one of the disks suffers a sector or whole-disk failure, lost data blocks can be reconstructed using the corresponding data and parity blocks from the other disks. Note that because the system already knows which disk has failed, parity is sufficient for error correction, not just error detection. For example, if the disk containing block data_0 fails, the block can be constructed by computing the exclusive-or of the parity block and the remaining data blocks:

$$\text{data}_0 = \text{parity} \oplus \text{data}_1 \oplus \dots \oplus \text{data}_{G-1}$$

To maximize performance, rotating parity RAIDs carefully organize their data layout by rotating parity and striping data to balance parallelism and sequential access:

- **Rotating parity.** Because the parity for a given set of blocks must be updated each time any of the data blocks are updated, the average parity block tends to be accessed more often than the average data block. To balance load, rather than having $G - 1$ disks store only data blocks and 1 disk store only parity blocks, each disk dedicates $\frac{1}{G}$ th of its space to parity and is responsible for storing $\frac{1}{G}$ th of the parity blocks and $\frac{G-1}{G}$ of the data blocks.
- **Striping data.** To balance parallelism versus sequential-access efficiency, a *strip* of several sequential blocks is placed on one disk before shifting to another disk for the next strip. A set of $G - 1$ data strips and their parity strip is called a *stripe*.

By striping data, requests larger than a block but smaller than a strip require just one disk to seek and then read or write the full sequential run of data rather than requiring multiple disks to seek and then read smaller sequential runs. Conversely, the RAID can service more widely spaced requests in parallel.

Definition: **strip**

Definition: **stripe**

Combining rotating parity and striping, we have the arrangement shown in Figure 14.6.

Example: Updating a RAID with rotating parity.

Question: For the rotating parity RAID in Figure 14.6, suppose you update data block 21. What disk I/O operations must occur?

Answer: The challenge is that we must not only update data block 21, we must also update the corresponding parity block. Since data block 21 is block 1 of its strip and the strip is part of stripe 1, we need to update parity block 1 of the parity strip for stripe 1 (*Parity (1,1,1)* in the figure).

It takes 4 I/O operations to update both the data and parity. First we read the old data D_{21} and parity $P_{1,1,1}$ and “remove” the old data from the parity calculation $P_{tmp} = P_{1,1,1} \oplus D_{21}$. Then we can compute the new parity from the new data $P'_{1,1,1} = P_{tmp} \oplus D'_{21}$. Finally we can write the new data D'_{21} and parity $P'_{1,1,1}$ to disks 2 and 1, respectively.

RAIDs with rotating parity have high overheads for small writes. Their overheads are far smaller for reads and for full-stripe writes.

Recovery. In either RAID arrangement, if a disk suffers a sector failure, the disk reports an error when there is an attempt to read the sector and, if necessary, remaps the damaged sector to a spare one. Then, the RAID system reconstructs the lost sector from the other disk(s) and rewrites it to the original disk.

If a disk suffers a whole-disk failure, an operator replaces the failed disk, and the RAID system reconstructs all of the disk's data from the other disk(s) and rewrites the data to the replacement disk. The average time from when a disk fails until it has been replaced and rewritten is called the *mean time to repair* (*MTTR*).
Definition: **mean time to repair** (*MTTR*)

Definition: **MTTR**

RAID reliability

A RAID with one redundant disk per group (e.g., mirroring or rotating parity RAIDs) can lose data in three ways: two full disk failures, a full disk failure and one or more sector failures on other disks, and overlapping sector failures on multiple disks. The expected time until one of these events occurs is called the *mean time to data loss* (*MTTDL*).

Definition: **mean time to data loss**

Definition: **MTTDL**

Two full-disk failures. If two disks fail, the system will be unable to reconstruct the missing data.

To get a sense of how serious a problem this might be, suppose that a system has N disks with one parity block per G blocks, and suppose that disks fail independently with a mean time to failure of *MTTF* and a mean time to replace a failed disk and recover its data of *MTTR*.

Then, when the system is operating properly, the expected time until the first failure is $MTTF/N$. Assuming $MTTR \ll MTTF$, there is essentially a race to replace the disk and reconstruct its data before a second disk fails. We lose this race and hit the second failure before the repair is done with probability $\frac{MTTF/(G-1)}{MTTR}$, giving us a mean time to data loss from multiple full-disk failures of

$$MTTDL_{two-full-disk} = \frac{MTTF^2}{N(G-1)MTTR}$$

Example: Mean time to double-disk failure.

Question: Suppose you have 100 disks organized into groups of 10, with one disk storing a parity block per nine disks storing data blocks. Assuming that disk failures are independent and the per-disk mean time to failure is 10^6 hours and assuming that the mean time to repair a failed disk is 10 hours, estimate the expected mean time to data loss due to a double-disk failure.

Answer: Because failures are assumed to occur independently and at a constant rate, we can use the equation above:

$$\begin{aligned}
 MTTDL_{two-full-disk} &= \frac{MTTF^2}{N(G-1)MTTR} \\
 &= \frac{(10^6\text{hours})^2}{(10^2)(9)(10\text{hours})} \\
 &\approx 10^8\text{hours}
 \end{aligned}$$

So, assuming independent failures at the expected rate and assuming no other sources of data loss, this organization appears to have raised the mean time to data loss from about 100 years (for a single disk) to about 10,000 years (for 90 disks worth of data and 10 disks worth of parity).

One full-disk failure and a sector failure. If one or more disks suffer sector failures and another disk suffers a full-disk failure, the RAID system can not recover all of its data. Assuming independent failures that arrive at a constant rate, we can estimate the mean time to data loss from this failure mode based on the expected time between full disk failures divided by the odds of failing to read all data needed to reconstruct the lost disk's data.

$$MTTDL_{disk+sector} = \frac{MTTF}{N} \cdot \frac{1}{P_{fail_recovery_read}}$$

Example: Mean time to failed disk and failed sector.

Question: Assuming that during recovery, latent sector errors are discovered at a rate of 1 per 10^{15} bits read and assuming that the mean time to failure for each of 100 1 TB disks organized into groups of 10 is 10^6 hours, what is the expected mean time to data loss due to full-disk failure combined with a sector failure?

Answer:

$$\begin{aligned}
 MTTDL_{disk+sector} &= \frac{MTTF}{N} \cdot \frac{1}{P_{fail_recovery_read}} \\
 &= \frac{10^6}{100} \cdot \frac{1}{P_{fail_recovery_read}}
 \end{aligned}$$

To estimate $P_{fail_recovery_read}$ we will assume that each bit fails independently and is successfully read with probability $(1/(1 - 10^{15}))$. Then the probability of reading 1 TB from each of 9 disks is

$$\begin{aligned} P_{\text{succeed_recovery_read}} &= (1/(1 - 10^{15}))^{9 \text{ disks} \cdot 10^{12} \frac{\text{bytes}}{\text{disk}} * 8 \frac{\text{bits}}{\text{byte}}} \\ &\approx 0.9306 \end{aligned}$$

So, there is nearly a 7% chance that recovery will fail, and we have

$$MTTDL_{\text{disk+sector}} = \frac{10^6}{100} \cdot \frac{1}{1 - 0.9306} = 1.44 \cdot 10^5 \text{ hours}$$

Notice that this rate of data loss is higher than the rate from double disk failures calculated above. Of course, the relative contributions of each failure mode will depend on disks' MTTF, size, and bit error rates as well as the system's MTTR.

Failure of two sectors sharing a redundant sector. In principle, it is also possible to lose data because the corresponding sectors fail on different disks. However, with billions of distinct sectors on each disk and small numbers of latent failures per disk, this failure mode is likely to be a negligible risk for most systems.

Overall data loss rate. If we assume independent failures and constant failure rates, then we can add the failure rates from the two significant failure modes to estimate the combined failure rate:

$$\begin{aligned} FailureRate_{\text{indep+const}} &= FailureRate_{\text{two-full-disk}} + FailureRate_{\text{disk+sector}} \\ &= \frac{1}{MTTDL_{\text{two-full-disk}}} + \frac{1}{MTTDL_{\text{disk+sector}}} \\ &= \frac{N(G-1)MTTR}{MTTF^2} + \frac{N \cdot P_{\text{fail_recovery_read}}}{MTTF} \\ &= \frac{N}{MTTF} \left(\frac{MTTR(G-1)}{MTTF} + P_{\text{fail_recovery_read}} \right) \end{aligned}$$

The total failure rate is thus the rate that the first disk fails times the rate that either a second disk in the group fails before the repair is completed or a sector error is encountered when the disks are being read to rebuild the lost disk.

We label the above $FailureRate_{\text{indep+const}}$ to emphasize the strong assumptions of independent failures and constant failure rates. As noted above, failures are likely to be correlated in many environments and failure rates of some devices

may increase over time. Both of these factors may result significantly higher failure rates than expected.

Example: Combined failure rate.

Question: For the system described in the previous examples (100 disks, rotating parity with a group size of 10, mean time to failure of 10^6 hours, mean time to repair of 10 hours, and nonrecoverable read error rate of one sector per 10^{15} bits) assuming that all failures are independent, estimate the MTTDL when both double-disk and single-disk-and-sector failures are considered.

Answer:

$$\begin{aligned} FailureRate_{indep+const} &= \frac{N}{MTTF} \left(\frac{MTTR(G-1)}{MTTF} + P_{fail_recovery_read} \right) \\ &= \frac{100 \text{ disks}}{10^6 \text{ hours}} \left(\frac{10 \text{ hours}}{10^6 \text{ hours}} + 0.0694 \right) \\ &= \frac{1}{10^4 \text{ hours}} \left(\frac{1}{10^4} + 0.0694 \right) \\ &= \frac{1}{10^4 \text{ hours}} (0.0695) \\ &= 6.95 \cdot 10^{-6} \frac{\text{failures}}{\text{hour}} \\ &= \end{aligned}$$

Inverting the failure rate gives the mean time to data loss:

$$\begin{aligned} MTTDL_{const+indep} &= \frac{1}{FailureRate_{indep+const}} \\ &= \frac{1}{1.44 \cdot 10^5 \frac{\text{hours}}{\text{failure}}} \\ &= 16.4 \frac{\text{years}}{\text{failure}} \end{aligned}$$

Two things in the example above are worth special note. First, for these parameters, the dominant cause of data loss is likely to be a single disk failure combined with a nonrecoverable read error during recovery. Second, for these parameters and this configuration, the resulting 6% chance of losing data per year may be unacceptable for many environments. As a result, systems use various techniques to improve the MTTDL in RAID systems.

Improving RAID reliability

What can be done to further improve reliability? Broadly speaking, we can do three things: (1) increase redundancy, (2) reduce nonrecoverable read error

rates, and (3) reduce mean time to repair. All of these approaches, in various combinations, are used in practice.

Here are some common approaches:

Increasing redundancy with more redundant disks. Rather than having a single redundant block per group (e.g., using two mirrored disks or using one parity disk for each stripe) systems can use double redundancy (e.g., three disk replicas or two error correction disks for each stripe.) In some cases, systems may use even more redundancy. For example, the Google File System (GFS) is designed to provide highly reliable and available storage across thousands of disks; by default GFS stores each data block on three different disks.

Definition: dual redundancy array A *dual redundancy array* is sometimes called *RAID 6*. To ensure that data can be reconstructed despite any two failures in a stripe, error blocks are generated using erasure codes such as Reed Solomon codes.

Definition: RAID 6 A system with dual redundancy can be much more reliable than a simple single redundancy RAID. With dual redundancy, the most likely data loss scenarios are (a) three full-disk failures or (b) a double-disk failure combined with one or more nonrecoverable read errors.

If we optimistically assume that failures are independent and occur at a constant rate, a system with two redundant disks per stripe has a potentially low combined data loss rate:

$$\text{FailureRate}_{\text{dual+indep+const}} = \frac{N}{\text{MTTF}} \frac{\text{MTTR}(G-1)}{\text{MTTF}} \left(\frac{\text{MTTR}(G-2)}{\text{MTTF}} + P_{\text{fail_recovery_read}} \right)$$

This data loss rate is nearly $\frac{\text{MTTF}}{\text{MTTR}(G-1)}$ times better than the single-parity data loss rate; for disks with MTTFs of over one million hours, MTTRs of under 10 hours, and groups sizes of ten or fewer disks, double parity improves the estimated rate by about a factor of 10,000.

We emphasize, however, that the above equation almost certainly underestimates the likely data loss rate for real systems, which may suffer correlated failures, varying failure rates, higher failure rates than advertised, and so on.

Reducing nonrecoverable read error rates with scrubbing. A storage device's sector-level error rates are typically expressed as a single *nonrecoverable read rates*, suggesting that the rate is constant. The reality is more complex. Depending on the device, errors may accumulate over time and heavier workloads may increase the rate that errors accumulate.

An important technique for reducing a disk's nonrecoverable read rate is **scrubbing**: periodically reading the entire contents of a disk, detecting sectors

with unrecoverable read errors, reconstructing the lost data from the remaining disks in the RAID array, and attempting to write and read the reconstructed data to and from the suspect sector. If writes and reads succeed, then the error was caused by a transient fault, and the disk continues to use the sector, but if the sector cannot be successfully accessed, the error is permanent, and the system remaps that sector to a spare and writes the reconstructed data there.

Reducing nonrecoverable read error rates with more reliable disks.

Different disk models promise significantly different nonrecoverable read error rates. In particular, in 2011, many disks aimed at laptops and personal computers claim unrecoverable read error rates of one per 10^{14} bits read, while disks aimed at enterprise servers often have lower storage densities but can promise unrecoverable read error rates of one per 10^{16} bits read. This two order of magnitude improvement greatly reduces the probability that a RAID system loses data from a combination of a full disk failure and a nonrecoverable read error during recovery.

Reducing mean time to repair with hot spares. Some systems include “hot spare” disk drives that are idle, but plugged into a server so that if one of the server’s disks fails, the hot spare can be automatically activated to replace the lost disk.

Note that even with hot spares, the mean time to repair a disk is limited by the time it takes to write the reconstructed data to it, and this time is often measured in hours. For example, if we have a 1 TB disk and can write at 100 MB/s, the mean time to repair for the disk will be at least 10^4 seconds—about 3 hours. In practice, repair time may be even larger if the bandwidth achieved is less than assumed here.

Reducing mean time to repair with declustering. Disks with hundreds of gigabytes to a few terabytes can take hours to fully write with reconstructed data. *Declustering* splits reconstruction of a failed disk across multiple disks. Definition: **declustering** Declustering thus allows parallel reconstruction, thus speeding up reconstruction and reducing MTTR.

For example, the Hadoop File System (HDFS) is a cluster file system that writes each data block to three out of potentially hundreds or thousands of disks. It chooses the three disks for each block more or less randomly. If one disk fails, it rereplicates the lost blocks approximately randomly across the remaining disks. If we have N disks each with a bandwidth of B , total reconstruction bandwidth can approach $\frac{N}{2}B$; for example, if there are 1000 disks with 100 MB/s bandwidths, reconstruction bandwidth can theoretically approach 500 GB/s, allowing rereplication of a 1 TB disk’s data in a few seconds.

In practice, rereplication will be slower than this for at least three reasons. First, resources other than the disk (e.g., the network) may bottleneck recovery.

Second, the system may throttle recovery speed to avoid starving user requests. Third, if a server crashes and its disks become inaccessible, the system may delay starting recovery—hoping that the server will soon recover—to avoid imposing extra load on the system.

Pitfalls

When constructing a reliable storage system, it is not enough to plug provide enough redundancy to tolerate a target number of failures. We also need to consider how failures are likely to occur (e.g., they may be correlated) and what it takes to correct them (e.g., successfully reading a lot of other data.) More specifically, be aware of the following pitfalls:

- **Assuming uncorrelated failures..** It is easy to get gaudy MTTDL numbers by adding a redundant device or two and multiplying the devices' MTTFs. But the simple equation on page 432 only applies when failures are uncorrelated. Even a 1% chance of correlated failures dramatically changes the estimate. Unfortunately, it is often difficult to estimate correlation rates *a priori*, so designers must sometimes just add a significant safety margin and hope that it is enough.
- **Ignoring the risk from latent errors..** It is not uncommon to see analyses of RAID reliability that considers full device failures but not nonrecoverable read failures. As we have seen above, nonrecoverable read errors can dramatically reduce the probability of successfully recovering data after a disk failure.
- **Not implementing scrubbing..** Periodically scrubbing disks to detect and correct latent errors can significantly reduce the risk of data loss. Although it can be difficult to predict the appropriate scrubbing frequency *a priori*, a system that uses scrubbing can monitor the rate at which noncorrectable read errors are found and corrected and use the measured rate to adjust the scrubbing frequency.
- **Not having a backup..** The techniques discussed in this section can protect a system against many, but not all, faults. For example, a widespread correlated failure (e.g., a building burning down), an operator error (e.g., “rm -r *”), or a software bug could corrupt or delete data stored across any number of redundant devices.

Definition: **backup**

A *backup* system provides storage that is separate from a system's main storage. Ideally, the separation is both physical and logical.

Definition: **physical separation**

Physical separation means that backup storage devices are in different locations than the primary storage devices. For example, some systems achieve physical separation by copying data to tape and storing the tapes in a different building than the main storage servers. Other systems

achieve physical separation by storing data to remote disk arrays such as those provided by cloud backup and disaster recovery services.

Logical separation means that the interface to the backup system is restricted to prevent premature deletion of data. For example, some backup systems provide an interface that allows a user to read *but not write* old versions of a file (e.g., the file as it existed one hour, two hours, four hours, one day, one week, one month, and one year ago.)

Definition: logical separation

14.2.3 Software integrity checks

Although storage devices include sector- or page-level checksums to detect data corruption, many recent file systems have included additional, higher-level, checksums and other integrity checks on their data.

These checks can catch a range of errors that hardware-level checksums can miss. For example, they can detect *wild writes* or *lost writes* where a bug in the operating system software, device driver software, or device firmware misdirects a write to the wrong block or page or fails to complete an intended write. They can also detect rare *ECC false negatives* when the hardware-level error correcting codes fail to detect a multi-bit corruption.

When a software integrity check fails on a block read or during latent-error scrubbing, the system reconstructs the lost or corrupted block using the redundant storage in the RAID.

Two examples of software integrity checks used today are *block integrity metadata* and *file system fingerprints*.

Block integrity metadata. Some file systems, like Network Appliance's WAFL file system, include *block integrity metadata* that allows the software to validate the results of each block it reads.

Definition: block integrity metadata

As Figure 14.7 illustrates, WAFL stores a 64 byte *data integrity segment* (DIS) with each 4 KB data block. The DIS contains a checksum of the data block, the identity of the data block (e.g., the ID of the file to which it belongs and the block's offset in that file), and a checksum of the DIS, itself.

Then, when a block is read, the system performs three checks. First, it checks the DIS's checksum. Second, it verifies that the data in the block corresponds to the checksum in the block's data integrity segment. Third, it verifies that the identity in the block's DIS corresponds to the file block it was intending to read. If all of these checks pass, the file system can be confident it is returning the correct data; if not, the file system can reconstruct the necessary data from redundant disks in the RAID.

Definition: **file system fingerprint** **File system fingerprints.** Some file systems, like Oracle's ZFS, include *file system fingerprints* that provide a checksum across the entire file system in a way that allows efficient checks and updates when individual blocks are read and written.

As illustrated in Figure 14.8 (a), all of ZFS's data structures are arranged in a tree of blocks with a root node called the *uberblock*. At each internal node of the tree, each reference to a child node includes both a pointer to and a checksum of the child. Thus, the reference to any subtree includes a checksum that covers all of that subtree's contents, and the *uberblock* holds a checksum that covers the entire file system.

When ZFS reads data (i.e., leaves of the tree) or metadata (i.e., internal nodes of the tree), it follows the pointers down the tree to find the right block to read, computing a checksum of each internal or leaf block and comparing it to the checksum stored with the block reference. Similarly, as Figure 14.8 (b) illustrates, when ZFS writes a block, it updates the references from the updated block to the *uberblock* so that each includes both the new checksum and (since ZFS never updates data structures in place) new block pointer.

Atomic update of data and parity

A challenge in implementing RAID is atomically updating both the data and the parity (or both data blocks in a RAID with mirroring.)

Consider what would happen if the RAID system in Figure 14.6 crashes in the middle updating block 21, after updating the data block on disk 2 but before updating the parity block on disk 1. Now, if disk 2 fails, the system will reconstruct the wrong (old) data for block 21.

The situation may be even worse if a write to a mirrored RAID is interrupted. Because reads can be serviced by either disk, reads of the inconsistent block may sometimes return the new value and sometimes return the old one.

Solutions. Three solutions and one non-solution are commonly used to solve (or not) the atomic update problem.

- **Nonvolatile write buffer.** Hardware RAID systems often include a battery-backed write buffer. An update is removed from the write buffer only once it is safely on disk. The RAID's startup procedures ensure that any data in the write buffer is written to disk after a crash or power outage.
 - **Transactional update.** RAID systems can use transactions to atomically update both the data block and the parity block. For example, Oracle's RAID-Z integrates RAID striping with the ZFS file system to avoid overwriting data in place and to atomically update data and parity.
 - **Recovery scan.** After a crash, the system can scan all of the blocks in the system and update any inconsistent parity blocks. Note that until that scan is complete, some parity blocks may be inconsistent, and incorrect data may be reconstructed if a disk fails. The Linux md (multiple device) software RAID driver uses this approach.
 - **Cross your fingers.** Some software and hardware RAID implementations do not ensure that the data and parity blocks are in sync after a crash. *Caveat emptor.*
-
-

RAID Levels

An early paper on RAIDs, “A Case for Redundant Arrays of Inexpensive Disks (RAID),” by David Patterson, Garth Gibson, and Randy Katz described a range of possible RAID organizations and named them RAID 0, RAID 1, RAID 2, RAID 3, RAID 4, and RAID 5. Several of these RAID levels were intended to illustrate key concepts rather than for real-world deployment.

Today, three of these variants are in wide use:

- **RAID 0—JBOD.** RAID level 0 spreads data across multiple disk without redundancy. Any disk failure results in data loss. For this reason, the term RAID is somewhat misleading, and this organization is often referred to as JBOD—Just a Bunch Of Disks.
- **RAID 1—Mirroring.** RAID level 1 mirrors identical data to two disks.
- **RAID 5—Rotating Parity.** RAID level 5 stripes data across G disks. $G - 1$ of the disks in a stripe store $G - 1$ different blocks of data and the remaining disk stores a parity block. The role of storing the parity block for different data blocks is rotated among the disks to balance load.

Subsequent to the “Case for RAID” paper, new organizations emerged, and many of them were named in the same spirit. Some of these names have become fairly standard.

- **RAID 6—Dual Redundancy.** RAID level 6 is similar to RAID level 5, but instead of one parity block per group, two redundant blocks are stored. These blocks are generated using erasure codes such as Reed Solomon codes that allow reconstruction of all of the original data as long as at most two disks fail.
- **RAID 10 and RAID 50—Nested RAID.** RAID 10 and RAID 50 were originally called RAID 1+0 and RAID 5+0. They simply combine RAID 0 with RAID 1 or RAID 5. For example, a RAID 10 system mirrors pairs of disks for redundancy (RAID 1), treats each pair of mirrored disks as a single reliable logical disk, and then stripes data non-redundantly across these logical disks (RAID 0).

Many other RAID levels have been proposed. In some cases, these new “levels” have more to do with marketing than technology (“Our company’s RAID 99+ is much better than your company’s puny RAID 14”). In any event, we regard the particular nomenclature used to describe exotic RAID organizations as relatively unimportant; our discussion focuses on mirroring (RAID 1), rotating parity (RAID 5), and dual redundancy (RAID 6). Other organizations can be analyzed using principles from these approaches.

Modeling Real Systems

The equations in the main text for estimating a system's mean time to data loss are only applicable if failure rates are constant and if failures are uncorrelated. Unfortunately, empirical studies often observe correlation among full-disk failures, among sector-level failures, and between sector-level and full-disk failures, and they frequently find failure rates that vary significantly with disks' ages. Unfortunately, if failure rates vary over time or failures are correlated, the failure arrival distribution is no longer described by an exponential distribution, and the math quickly gets difficult.

One solution is to use randomized simulation to estimate the probability of data loss over some duration of interest. For example, we might want to estimate the probability of losing data over 10 years for a 1000-disk system organized in groups of 10 disks with rotating parity.

To do this, our simulation would track which disks are functioning normally, which have latent sector errors, and which have suffered full disk failures. The transitions between states could be based on measurement studies or field data on key factors like (a) the rate that disks suffer full disk failures (possibly dependent on the disks' ages, the number of recent full disk failures, or the number of individual sector failures a disk has had), (b) the rate at which sector failures arise (possibly dependent on the age of the disk, workload of the disk, and recent frequency of sector failures), (c) the repair time when a disk fails, and (d) the expected time for scrubbing to detect and repair a sector error.

To estimate the probability of data loss, we would repeatedly simulate the system for a decade and count the number of times the system enters a state in which data is lost (i.e., a group has two full disk failures or has both a full disk failure and a sector failure on another disk.)

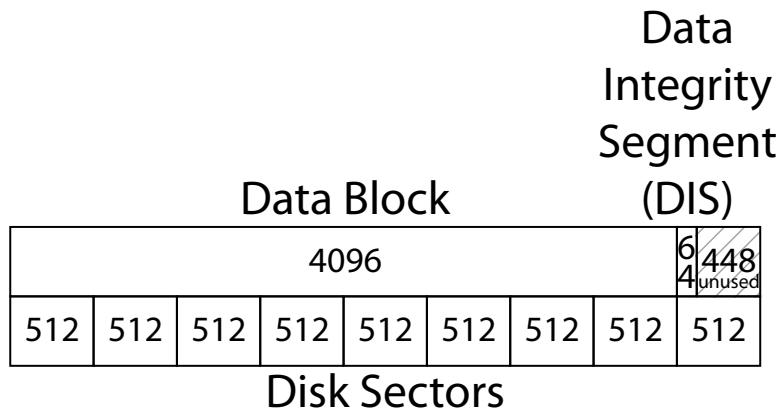


Figure 14.7: To improve reliability Network Appliance's WAFL file system stores a 64 byte data integrity segment (DIS) with each 4 KB data block.

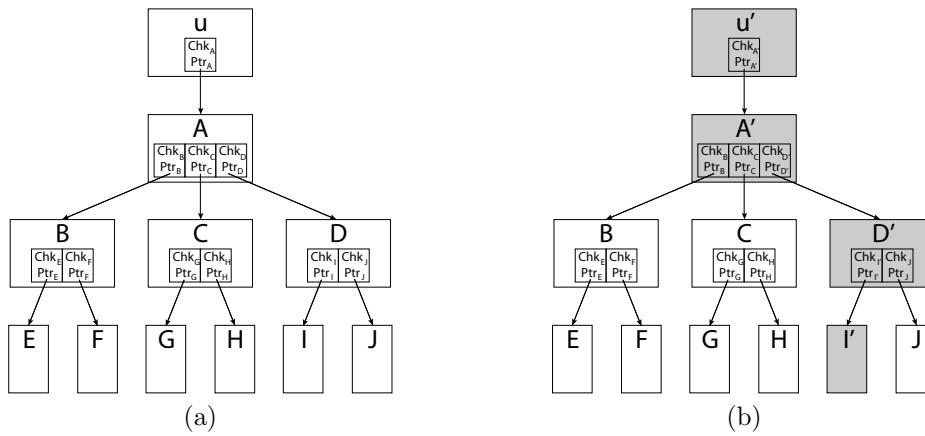


Figure 14.8: (a) ZFS stores all data in a Merkle tree so that each node of the tree includes both a pointer to and a checksum of each of its children (Chk and Ptr in the figure). On an update (b) all nodes from the updated block (I') to the root (u') are updated to reflect the new pointer and checksum values.

Layers Upon Layers Upon Layers

In this chapter we focus on error detection and correction at three levels: the individual storage devices (e.g., disks and flash), storage architectures (e.g., RAID), and file systems.

Today, storage systems with important data often include not just these layers, but additional ones. Enterprise and cloud storage systems distribute data across several geographically-distributed sites and may include high-level checksums on that geographically-replicated data. Within a site, they may replicate data across multiple servers using what is effectively a distributed file system. At each server, the distributed file system may store data using a local file system that includes file-system-level checksums on the locally-stored data. And, invariably, the local server will use storage devices that detect and sometimes correct low-level errors.

Although we do not discuss cross-machine and geographic replication in any detail, the principles described in this chapter also apply to these systems.

Exercises

5. Go to an on-line site that sells hard disk drives, and find the largest capacity disk you can buy for less than \$200. Now, track down the spec sheet for the disk and, given the disk's specified bit error rate (or unrecoverable read rate), estimate the probability of encountering an error if you read every sector on the disk once.
6. Suppose we define a RAID's *access cost* as the number disk accesses divided by the number of data blocks read or written. For each of following configurations and workloads, what is the access cost?
 - (a) Workload: a series of random 1-block writes
Configuration: mirroring
 - (b) Workload: a series of random 1-block writes
Configuration: distributed parity
 - (c) Workload: a series of random 1-block reads
Configuration: mirroring
 - (d) Workload: a series of random 1-block reads
Configuration: distributed parity
 - (e) Workload: a series of random 1-block reads
Configuration: distributed parity with groupsize G and one failed disk
 - (f) Workload: a long sequential write
Configuration: mirroring
 - (g) Workload: a long sequential write
Configuration: distributed parity with a group size of G
7. Suppose that an engineer who has not taken this class tries to create a disk array with dual-redundancy but instead of using an appropriate error correcting code such as Reed Solomon, the engineer simply stores a copy of each parity block on two disks. e.g.,

data0	data1	data2	data3	parity	parity
-------	-------	-------	-------	--------	--------

 Give an example of how a two-disk failure can cause a stripe to lose data in such a system. Explain why data cannot be reconstructed in that case.
8. Some RAID systems improve reliability with intra-disk redundancy to protect against nonrecoverable read failures. For example, each individual disk on such a system might reserve one 4KB parity block in every 32 KB extent and then store 28KB (7 4KB blocks) of data and 4 KB (1 4KB block) of parity in each extent.

In this arrangement, each data block is protected by two parity blocks: one interdisk parity block on a different disk and an intradisk parity block on the same disk.

This approach may reduce a disk's effective nonrecoverable read error rate because if one block in an extent is lost, it can be recovered from the remaining sectors and parity on the disk. Of course, if multiple blocks in the same extent are lost, the system must rely on redundancy from other disks.

- (a) Assuming that a disk's nonrecoverable read errors are independent and occur at a rate of one lost 512 byte sector per 10^{15} bits read, what is the effective nonrecoverable read error rate if the operating system stores one parity block per seven data blocks on the disk?

Hint: You may find the `bc` or `dc` arbitrary-precision calculators useful. These programs are standard in many Unix, Linux, and OSX distributions. See the man pages for instructions.

- (b) Why is the above likely to significantly overstate the impact of intra-disk redundancy?

9. Many RAID implementations allow on-line repair in which the system continues to operate after a disk failure, while a new empty disk is inserted to replace the failed disk, and while regenerating and copying data to the new disk.

Sketch a design for a 2-disk, mirrored RAID that allows the system to remain on-line during reconstruction, while still ensuring that when the data copying is done, the new disk is properly reconstructed (i.e., it is an exact copy of other disk.)

In particular, specify (1) what is done by a recovery thread, (2) what is done on a read during recovery, and (3) what is done on a write during recovery. Also explain why your system will operate correctly even if a crash occurs in the middle of reconstruction.

10. Suppose you are willing to sacrifice no more than 1% of a disk's bandwidth to scrubbing. What is maximum frequency at which you could scrub a 1 TB disk with 100 MB/s bandwidth?

11. Suppose a 3 TB disk in a mirrored RAID system crashes. Assuming the disks used in the system can sustain 100MB/s sequential bandwidth, what is the minimum mean time to repair that can be achieved? Why might a system be configured to perform recovery slower than this?

14.3 Conclusion and future directions

Although individual storage devices include internal error correcting codes, additional redundancy for error detection and correction is often needed to provide acceptably reliable storage. In fact, today, it is seldom acceptable to store valuable data on a single device without some form of RAID-style redundancy. By the same token, many if not most file systems designed over the past decade have included software error checking to catch data corruption and loss occurrences that are not detectable by device-level hardware checks.

Increasingly now and in the future, systems go beyond just replicating data across multiple disks on a single server to distributed replication across multiple servers. Sometimes these replicas are configured to protect data even if significant physical disasters occur.

For example, Amazon's Simple Storage Service (S3) is a cloud storage service that allows customers to pay a monthly fee to store data on servers run by Amazon. As of when this paragraph was written (January 2012), Amazon stated that the system was "designed to provide 99.99999999% durability ... of objects over a given year." To provide such high reliability, Amazon must protect against disasters like a data center being destroyed in a fire. Amazon S3 therefore stores data at multiple data centers, it works to quickly repair lost redundancy, and it periodically scans stored data to verify its integrity via software checksums.

Exercises

1. Suppose that a text editor application uses the `rename` technique discussed on page 403 for safely saving updates by saving the updated file to a new file (e.g., `#doc.txt#` and then calling `rename(''#doc.txt'', ''doc.txt'')`) to change the name of the updated file from `#doc.txt#` to `doc.txt`. Posix `rename` promises that the update to `doc.txt` will be atomic—even if a crash occurs, `doc.txt` will refer to either the old file or the new one. However, Posix does not guarantee that the entire rename operation will be atomic. In particular, Posix allows implementations in which there is a window in which a crash could result in a state where both `doc.txt` and `#doc.txt#` refer to the same, new document.
 - a. How should a text-editing application react if, on startup, it sees both `doc.txt` and `doc.txt` and (i) both refer the same file or (ii) each refers to a file with different contents?
 - b. Why might Posix permit this corner case (where we may end up with two names that refer to the same file) to exist?
 - c. Explain how an FFS-based file system without transactions could use the "ad hoc" approach discussed in Section 14.1.1 to ensure that (i)

`doc.txt` always refers to either the old or new file, (ii) the new file is never lost – it is always available as at least one of `doc.txt` or `#doc.txt#`, and (iii) there is some window where the new file may be accessed as both `doc.txt` and `#doc.txt#`.

- d. Section 14.1.1 discusses three reasons that few modern file systems use the “ad-hoc” approach. However, many text editors still do something like this. Why have the three issues had less effect on applications like text editors than on file systems?

- 2. Above, we defined two-phase locking for basic mutual exclusion locks. Extend the definition of two-phase locking for systems that use readers-writers locks.

- 3. Suppose that x and y represent the number of hours two managers have assigned you to work on each of two tasks with a constraint that $x + y \leq 40$. On page 414, we showed that snapshot isolation could allow one transaction to update x and another concurrent transaction to update y in a way that would violate the constraint $x + y \leq 40$. Is such an anomaly possible under serializability? Why or why not?

- 4. Suppose you have transactional storage system `tStore` that allows you to read and write fixed-sized 2048-byte blocks of data within transactions, and you run the following code.

```

...
byte b1[2048];  byte b2[2048];
byte b3[2048];  byte b4[2048];

TransID t1 = tStore.beginTransaction();
TransID t2 = tStore.beginTransaction();
TransID t3 = tStore.beginTransaction();
TransID t4 = tStore.beginTransaction();

// Interface is
//      writeBlock(TransID tid, int blockNum, byte buffer[]);
tStore.writeBlock(t1, 1, ALL_ONES);
tStore.writeBlock(t1, 2, ALL_TWOS);
tStore.writeBlock(t2, 3, ALL_THREES);
tStore.writeBlock(t1, 3, ALL_FOURS);
tStore.writeBlock(t1, 2, ALL_FIVES);
tStore.writeBlock(t3, 2, ALL_SIXES);
tStore.writeBlock(t4, 4, ALL_SEVENS);
tStore.readBlock(t2, 1, b1);
tStore.commit(t3);
tStore.readBlock(t2, 3, b2);
tStore.commit(t2);
tStore.readBlock(t1, 3, b3);
tStore.readBlock(t4, 3, b4);
tStore.commit(t1);

// At this point, the system crashes

```

The system crashes at the point indicated above.

Assume that `ALL_ONES`, `ALL_TWOS`, etc. are each arrays of 2048 bytes with the indicated value. Assume that when the program is started, all blocks in the `tStore` have the value `ALL_ZEROS`.

Just before the system crashes, what is the value of `b1` and what is the value of `b2`?

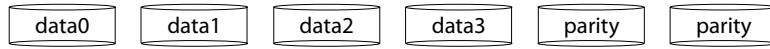
- (a) In the program above, just before the system crashes, what is the value of `b3` and what is the value of `b4`?
- (b) Suppose that after the program above runs and crashes at the indicated point. After the system restarts and completes recovery and all write-backs, what are the values stored in each of blocks 1, 2, 3, 4, and 5 of the `tStore`?

PROBLEM:

```
[[ compare performance of 1000 updates in place v. transaction --
  fall 2011 exam ]]
```

5. Go to an on-line site that sells hard disk drives, and find the largest capacity disk you can buy for less than \$200. Now, track down the spec sheet for the disk and, given the disk's specified bit error rate (or unrecoverable read rate), estimate the probability of encountering an error if you read every sector on the disk once.
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 - (g) Workload: a long sequential write
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7. Suppose that an engineer who has not taken this class tries to create a disk array with dual-redundancy but instead of using an appropriate error correcting code such as Reed Solomon, the engineer simply stores a copy of each parity block on two disks. e.g.,



Give an example of how a two-disk failure can cause a stripe to lose data in such a system. Explain why data cannot be reconstructed in that case.

8. Some RAID systems improve reliability with intra-disk redundancy to protect against nonrecoverable read failures. For example, each individual disk on such a system might reserve one 4KB parity block in every 32 KB extent and then store 28KB (7 4KB blocks) of data and 4 KB (1 4KB block) of parity in each extent.

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- (b) Why is the above likely to significantly overstate the impact of intra-disk redundancy?

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11. Suppose a 3 TB disk in a mirrored RAID system crashes. Assuming the disks used in the system can sustain 100MB/s sequential bandwidth, what is the minimum mean time to repair that can be achieved? Why might a system be configured to perform recovery slower than this?

Part V

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