Part III

Persistence

35

A Dialogue on Persistence

Professor: *And thus we reach the third of our four ... err... three pillars of operating systems:* ***persistence****.*

Student: *Did you say there were three pillars, or four? What is the fourth?*

Professor: *No. Just three, young student, just three. Trying to keep it simple here.*

Student: *OK, fine. But what is persistence, oh fine and noble professor?*

Professor: *Actually, you probably know what it means in the traditional sense, right? As the dictionary would say: "a firm or obstinate continuance in a course of action in spite of difficulty or opposition."*

Student: *It's kind of like taking your class: some obstinance required.*

Professor: *Ha! Yes. But persistence here means something else. Let me explain. Imagine you are outside, in a field, and you pick a —*

Student: *(interrupting) I know! A peach! From a peach tree!*

Professor: *I was going to say apple, from an apple tree. Oh well; we'll do it your way, I guess.*

Student: *(stares blankly)*

Professor: *Anyhow, you pick a peach; in fact, you pick many many peaches, but you want to make them last for a long time. Winter is hard and cruel in Wisconsin, after all. What do you do?*

Student: *Well, I think there are some different things you can do. You can pickle it! Or bake a pie. Or make a jam of some kind. Lots of fun!*

Professor: *Fun? Well, maybe. Certainly, you have to do a lot more work to make the peach* ***persist****. And so it is with information as well; making information persist, despite computer crashes, disk failures, or power outages is a tough and interesting challenge.*

Student: *Nice segue; you're getting quite good at that.*

Professor: *Thanks! A professor can always use a few kind words, you know.*

Student: *I'll try to remember that. I guess it's time to stop talking peaches, and start talking computers?*

Professor: *Yes, it is that time...*

36

I/O Devices

Before delving into the main content of this part of the book (on persistence), we first introduce the concept of an **input/output (I/O) device** and show how the operating system might interact with such an entity. I/O is quite critical to computer systems, of course; imagine a program without any input (it produces the same result each time); now imagine a program with no output (what was the purpose of it running?). Clearly, for computer systems to be interesting, both input and output are required. And thus, our general problem:

CRUX: HOW TO INTEGRATE I/O INTO SYSTEMS
How should I/O be integrated into systems? What are the general mechanisms? How can we make them efficient?

36.1 System Architecture

To begin our discussion, let's look at the structure of a typical system (Figure 36.1). The picture shows a single CPU attached to the main memory of the system via some kind of **memory bus** or interconnect. Some devices are connected to the system via a general **I/O bus**, which in many modern systems would be **PCI** (or one of its many derivatives); graphics and some other higher-performance I/O devices might be found here. Finally, even lower down are one or more of what we call a **peripheral bus**, such as **SCSI**, **SATA**, or **USB**. These connect the slowest devices to the system, including **disks**, **mice**, and other similar components.

One question you might ask is: why do we need a hierarchical structure like this? Put simply: physics, and cost. The faster a bus is, the shorter it must be; thus, a high-performance memory bus does not have much room to plug devices and such into it. In addition, engineering a bus for high performance is quite costly. Thus, system designers have adopted this hierarchical approach, where components that demand high performance (such as the graphics card) are nearer the CPU. Lower per-

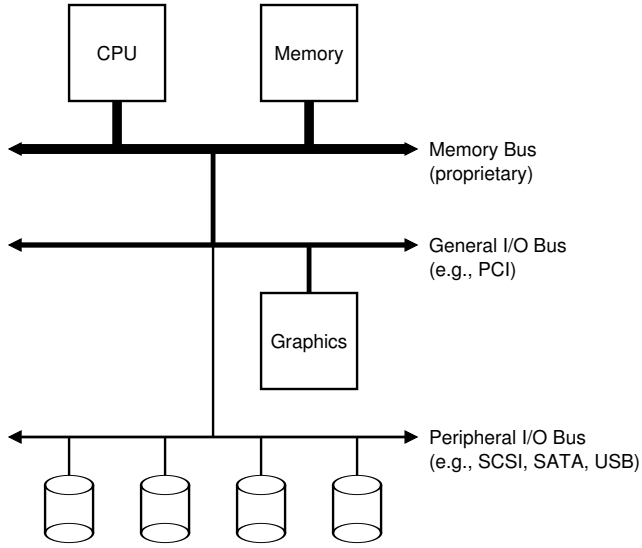


Figure 36.1: **Prototypical System Architecture**

formance components are further away. The benefits of placing disks and other slow devices on a peripheral bus are manifold; in particular, you can place a large number of devices on it.

36.2 A Canonical Device

Let us now look at a canonical device (not a real one), and use this device to drive our understanding of some of the machinery required to make device interaction efficient. From Figure 36.2, we can see that a device has two important components. The first is the hardware **interface** it presents to the rest of the system. Just like a piece of software, hardware must also present some kind of interface that allows the system software to control its operation. Thus, all devices have some specified interface and protocol for typical interaction.

The second part of any device is its **internal structure**. This part of the device is implementation specific and is responsible for implementing the abstraction the device presents to the system. Very simple devices will have one or a few hardware chips to implement their functionality; more complex devices will include a simple CPU, some general purpose memory, and other device-specific chips to get their job done. For example, modern RAID controllers might consist of hundreds of thousands of lines of **firmware** (i.e., software within a hardware device) to implement its functionality.

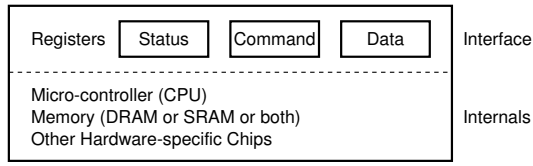


Figure 36.2: **A Canonical Device**

36.3 The Canonical Protocol

In the picture above, the (simplified) device interface is comprised of three registers: a **status** register, which can be read to see the current status of the device; a **command** register, to tell the device to perform a certain task; and a **data** register to pass data to the device, or get data from the device. By reading and writing these registers, the operating system can control device behavior.

Let us now describe a typical interaction that the OS might have with the device in order to get the device to do something on its behalf. The protocol is as follows:

```
While (STATUS == BUSY)
    ; // wait until device is not busy
Write data to DATA register
Write command to COMMAND register
    (Doing so starts the device and executes the command)
While (STATUS == BUSY)
    ; // wait until device is done with your request
```

The protocol has four steps. In the first, the OS waits until the device is ready to receive a command by repeatedly reading the status register; we call this **polling** the device (basically, just asking it what is going on). Second, the OS sends some data down to the data register; one can imagine that if this were a disk, for example, that multiple writes would need to take place to transfer a disk block (say 4KB) to the device. When the main CPU is involved with the data movement (as in this example protocol), we refer to it as **programmed I/O (PIO)**. Third, the OS writes a command to the command register; doing so implicitly lets the device know that both the data is present and that it should begin working on the command. Finally, the OS waits for the device to finish by again polling it in a loop, waiting to see if it is finished (it may then get an error code to indicate success or failure).

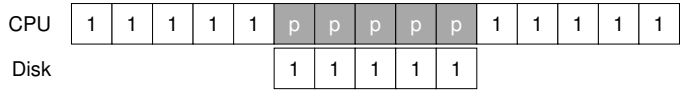
This basic protocol has the positive aspect of being simple and working. However, there are some inefficiencies and inconveniences involved. The first problem you might notice in the protocol is that polling seems inefficient; specifically, it wastes a great deal of CPU time just waiting for the (potentially slow) device to complete its activity, instead of switching to another ready process and thus better utilizing the CPU.

THE CRUX: HOW TO AVOID THE COSTS OF POLLING
How can the OS check device status without frequent polling, and thus lower the CPU overhead required to manage the device?

36.4 Lowering CPU Overhead With Interrupts

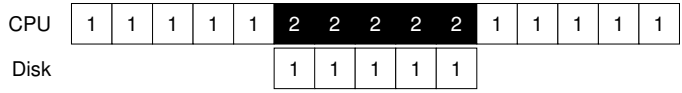
The invention that many engineers came upon years ago to improve this interaction is something we've seen already: the **interrupt**. Instead of polling the device repeatedly, the OS can issue a request, put the calling process to sleep, and context switch to another task. When the device is finally finished with the operation, it will raise a hardware interrupt, causing the CPU to jump into the OS at a pre-determined **interrupt service routine (ISR)** or more simply an **interrupt handler**. The handler is just a piece of operating system code that will finish the request (for example, by reading data and perhaps an error code from the device) and wake the process waiting for the I/O, which can then proceed as desired.

Interrupts thus allow for **overlap** of computation and I/O, which is key for improved utilization. This timeline shows the problem:



In the diagram, Process 1 runs on the CPU for some time (indicated by a repeated `1` on the CPU line), and then issues an I/O request to the disk to read some data. Without interrupts, the system simply spins, polling the status of the device repeatedly until the I/O is complete (indicated by a `p`). The disk services the request and finally Process 1 can run again.

If instead we utilize interrupts and allow for overlap, the OS can do something else while waiting for the disk:



In this example, the OS runs Process 2 on the CPU while the disk services Process 1's request. When the disk request is finished, an interrupt occurs, and the OS wakes up Process 1 and runs it again. Thus, *both* the CPU and the disk are properly utilized during the middle stretch of time.

Note that using interrupts is not *always* the best solution. For example, imagine a device that performs its tasks very quickly: the first poll usually finds the device to be done with task. Using an interrupt in this case will actually *slow down* the system: switching to another process, handling the interrupt, and switching back to the issuing process is expensive. Thus, if a device is fast, it may be best to poll; if it is slow, interrupts, which allow

TIP: INTERRUPTS NOT ALWAYS BETTER THAN PIO
Although interrupts allow for overlap of computation and I/O, they only really make sense for slow devices. Otherwise, the cost of interrupt handling and context switching may outweigh the benefits interrupts provide. There are also cases where a flood of interrupts may overload a system and lead it to livelock [MR96]; in such cases, polling provides more control to the OS in its scheduling and thus is again useful.

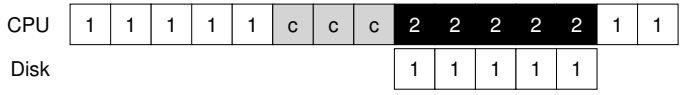
overlap, are best. If the speed of the device is not known, or sometimes fast and sometimes slow, it may be best to use a **hybrid** that polls for a little while and then, if the device is not yet finished, uses interrupts. This **two-phased** approach may achieve the best of both worlds.

Another reason not to use interrupts arises in networks [MR96]. When a huge stream of incoming packets each generate an interrupt, it is possible for the OS to **livelock**, that is, find itself only processing interrupts and never allowing a user-level process to run and actually service the requests. For example, imagine a web server that suddenly experiences a high load due to the "slashdot effect". In this case, it is better to occasionally use polling to better control what is happening in the system and allow the web server to service some requests before going back to the device to check for more packet arrivals.

Another interrupt-based optimization is **coalescing**. In such a setup, a device which needs to raise an interrupt first waits for a bit before delivering the interrupt to the CPU. While waiting, other requests may soon complete, and thus multiple interrupts can be coalesced into a single interrupt delivery, thus lowering the overhead of interrupt processing. Of course, waiting too long will increase the latency of a request, a common trade-off in systems. See Ahmad et al. [A+11] for an excellent summary.

36.5 More Efficient Data Movement With DMA

Unfortunately, there is one other aspect of our canonical protocol that requires our attention. In particular, when using programmed I/O (PIO) to transfer a large chunk of data to a device, the CPU is once again overburdened with a rather trivial task, and thus wastes a lot of time and effort that could better be spent running other processes. This timeline illustrates the problem:

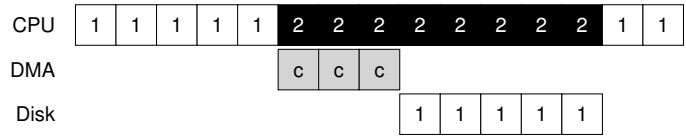


In the timeline, Process 1 is running and then wishes to write some data to the disk. It then initiates the I/O, which must copy the data from memory to the device explicitly, one word at a time (marked `c` in the diagram). When the copy is complete, the I/O begins on the disk and the CPU can finally be used for something else.

THE CRUX: HOW TO LOWER PIO OVERHEADS
With PIO, the CPU spends too much time moving data to and from devices by hand. How can we offload this work and thus allow the CPU to be more effectively utilized?

The solution to this problem is something we refer to as **Direct Memory Access (DMA)**. A DMA engine is essentially a very specific device within a system that can orchestrate transfers between devices and main memory without much CPU intervention.

DMA works as follows. To transfer data to the device, for example, the OS would program the DMA engine by telling it where the data lives in memory, how much data to copy, and which device to send it to. At that point, the OS is done with the transfer and can proceed with other work. When the DMA is complete, the DMA controller raises an interrupt, and the OS thus knows the transfer is complete. The revised timeline:



From the timeline, you can see that the copying of data is now handled by the DMA controller. Because the CPU is free during that time, the OS can do something else, here choosing to run Process 2. Process 2 thus gets to use more CPU before Process 1 runs again.

36.6 Methods Of Device Interaction

Now that we have some sense of the efficiency issues involved with performing I/O, there are a few other problems we need to handle to incorporate devices into modern systems. One problem you may have noticed thus far: we have not really said anything about how the OS actually communicates with the device! Thus, the problem:

THE CRUX: HOW TO COMMUNICATE WITH DEVICES
How should the hardware communicate with a device? Should there be explicit instructions? Or are there other ways to do it?

Over time, two primary methods of device communication have developed. The first, oldest method (used by IBM mainframes for many years) is to have explicit **I/O instructions**. These instructions specify a way for the OS to send data to specific device registers and thus allow the construction of the protocols described above.

For example, on x86, the `in` and `out` instructions can be used to communicate with devices. For example, to send data to a device, the caller specifies a register with the data in it, and a specific *port* which names the device. Executing the instruction leads to the desired behavior.

Such instructions are usually **privileged**. The OS controls devices, and the OS thus is the only entity allowed to directly communicate with them. Imagine if any program could read or write the disk, for example: total chaos (as always), as any user program could use such a loophole to gain complete control over the machine.

The second method to interact with devices is known as **memory-mapped I/O**. With this approach, the hardware makes device registers available as if they were memory locations. To access a particular register, the OS issues a load (to read) or store (to write) the address; the hardware then routes the load/store to the device instead of main memory.

There is not some great advantage to one approach or the other. The memory-mapped approach is nice in that no new instructions are needed to support it, but both approaches are still in use today.

36.7 Fitting Into The OS: The Device Driver

One final problem we will discuss: how to fit devices, each of which have very specific interfaces, into the OS, which we would like to keep as general as possible. For example, consider a file system. We'd like to build a file system that worked on top of SCSI disks, IDE disks, USB keychain drives, and so forth, and we'd like the file system to be relatively oblivious to all of the details of how to issue a read or write request to these difference types of drives. Thus, our problem:

THE CRUX: HOW TO BUILD A DEVICE-NEUTRAL OS
How can we keep most of the OS device-neutral, thus hiding the details of device interactions from major OS subsystems?

The problem is solved through the age-old technique of **abstraction**. At the lowest level, a piece of software in the OS must know in detail how a device works. We call this piece of software a **device driver**, and any specifics of device interaction are encapsulated within.

Let us see how this abstraction might help OS design and implementation by examining the Linux file system software stack. Figure 36.3 is a rough and approximate depiction of the Linux software organization. As you can see from the diagram, a file system (and certainly, an application above) is completely oblivious to the specifics of which disk class it is using; it simply issues block read and write requests to the generic block layer, which routes them to the appropriate device driver, which handles the details of issuing the specific request. Although simplified, the diagram shows how such detail can be hidden from most of the OS.

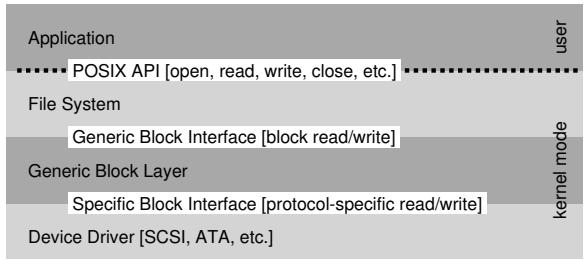


Figure 36.3: **The File System Stack**

Note that such encapsulation can have its downside as well. For example, if there is a device that has many special capabilities, but has to present a generic interface to the rest of the kernel, those special capabilities will go unused. This situation arises, for example, in Linux with SCSI devices, which have very rich error reporting; because other block devices (e.g., ATA/IDE) have much simpler error handling, all that higher levels of software ever receive is a generic `EIO` (generic IO error) error code; any extra detail that SCSI may have provided is thus lost to the file system [G08].

Interestingly, because device drivers are needed for any device you might plug into your system, over time they have come to represent a huge percentage of kernel code. Studies of the Linux kernel reveal that over 70% of OS code is found in device drivers [C01]; for Windows-based systems, it is likely quite high as well. Thus, when people tell you that the OS has millions of lines of code, what they are really saying is that the OS has millions of lines of device-driver code. Of course, for any given installation, most of that code may not be active (i.e., only a few devices are connected to the system at a time). Perhaps more depressingly, as drivers are often written by "amateurs" (instead of full-time kernel developers), they tend to have many more bugs and thus are a primary contributor to kernel crashes [S03].

36.8 Case Study: A Simple IDE Disk Driver

To dig a little deeper here, let's take a quick look at an actual device: an IDE disk drive [L94]. We summarize the protocol as described in this reference [W10]; we'll also peek at the xv6 source code for a simple example of a working IDE driver [CK+08].

An IDE disk presents a simple interface to the system, consisting of four types of register: control, command block, status, and error. These registers are available by reading or writing to specific "I/O addresses" (such as `0x3F6` below) using (on x86) the `in` and `out` I/O instructions.

```
Control Register:
  Address 0x3F6 = 0x08 (0000 1RE0): R=reset, E=0 means "enable interrupt"

Command Block Registers:
  Address 0x1F0 = Data Port
  Address 0x1F1 = Error
  Address 0x1F2 = Sector Count
  Address 0x1F3 = LBA low byte
  Address 0x1F4 = LBA mid byte
  Address 0x1F5 = LBA hi  byte
  Address 0x1F6 = 1B1D TOP4LBA: B=LBA, D=drive
  Address 0x1F7 = Command/status

Status Register (Address 0x1F7):
    7      6    5      4    3    2    1     0
   BUSY  READY FAULT SEEK  DRQ  CORR IDDEX ERROR

Error Register (Address 0x1F1): (check when Status ERROR==1)
    7      6    5      4    3    2    1    0
   BBK    UNC   MC   IDNF  MCR  ABRT T0NF AMNF

   BBK  = Bad Block
   UNC  = Uncorrectable data error
   MC   = Media Changed
   IDNF = ID mark Not Found
   MCR  = Media Change Requested
   ABRT = Command aborted
   T0NF = Track 0 Not Found
   AMNF = Address Mark Not Found
```

Figure 36.4: **The IDE Interface**

The basic protocol to interact with the device is as follows, assuming it has already been initialized.

- **Wait for drive to be ready.** Read Status Register (0x1F7) until drive is READY and not BUSY.
- **Write parameters to command registers.** Write the sector count, logical block address (LBA) of the sectors to be accessed, and drive number (master=0x00 or slave=0x10, as IDE permits just two drives) to command registers (0x1F2-0x1F6).
- **Start the I/O.** by issuing read/write to command register. Write READ—WRITE command to command register (0x1F7).
- **Data transfer (for writes):** Wait until drive status is READY and DRQ (drive request for data); write data to data port.
- **Handle interrupts.** In the simplest case, handle an interrupt for each sector transferred; more complex approaches allow batching and thus one final interrupt when the entire transfer is complete.
- **Error handling.** After each operation, read the status register. If the ERROR bit is on, read the error register for details.

Most of this protocol is found in the xv6 IDE driver (Figure 36.5), which (after initialization) works through four primary functions. The first is `ide_rw()`, which queues a request (if there are others pending), or issues it directly to the disk (via `ide_start_request()`); in either

```
static int ide_wait_ready() {
  while (((int r = inb(0x1f7)) & IDE_BSY) || !(r & IDE_DRDY))
    ;                               // loop until drive isn't busy
}

static void ide_start_request(struct buf *b) {
  ide_wait_ready();
  outb(0x3f6, 0);                   // generate interrupt
  outb(0x1f2, 1);                   // how many sectors?
  outb(0x1f3, b->sector & 0xff);    // LBA goes here ...
  outb(0x1f4, (b->sector >> 8) & 0xff);   // ... and here
  outb(0x1f5, (b->sector >> 16) & 0xff);  // ... and here!
  outb(0x1f6, 0xe0 | ((b->dev&1)<<4) | ((b->sector>>24)&0x0f));
  if(b->flags & B_DIRTY){
    outb(0x1f7, IDE_CMD_WRITE);     // this is a WRITE
    outsl(0x1f0, b->data, 512/4);   // transfer data too!
  } else {
    outb(0x1f7, IDE_CMD_READ);      // this is a READ (no data)
  }
}

void ide_rw(struct buf *b) {
  acquire(&ide_lock);
  for (struct buf **pp = &ide_queue; *pp; pp=&(*pp)->qnext)
    ;                               // walk queue
  *pp = b;                          // add request to end
  if (ide_queue == b)               // if q is empty
    ide_start_request(b);           // send req to disk
  while ((b->flags & (B_VALID|B_DIRTY)) != B_VALID)
    sleep(b, &ide_lock);            // wait for completion
  release(&ide_lock);
}

void ide_intr() {
  struct buf *b;
  acquire(&ide_lock);
  if (!(b->flags & B_DIRTY) && ide_wait_ready() >= 0)
    insl(0x1f0, b->data, 512/4);    // if READ: get data
  b->flags |= B_VALID;
  b->flags &= ~B_DIRTY;
  wakeup(b);                        // wake waiting process
  if ((ide_queue = b->qnext) != 0) // start next request
    ide_start_request(ide_queue);  // (if one exists)
  release(&ide_lock);
}
```

Figure 36.5: **The xv6 IDE Disk Driver (Simplified)**

case, the routine waits for the request to complete and the calling process is put to sleep. The second is `ide_start_request()`, which is used to send a request (and perhaps data, in the case of a write) to the disk; the `in` and `out` x86 instructions are called to read and write device registers, respectively. The start request routine uses the third function, `ide_wait_ready()`, to ensure the drive is ready before issuing a request to it. Finally, `ide_intr()` is invoked when an interrupt takes place; it reads data from the device (if the request is a read, not a write), wakes the process waiting for the I/O to complete, and (if there are more requests in the I/O queue), launches the next I/O via `ide_start_request()`.

36.9 Historical Notes

Before ending, we include a brief historical note on the origin of some of these fundamental ideas. If you are interested in learning more, read Smotherman's excellent summary [S08].

Interrupts are an ancient idea, existing on the earliest of machines. For example, the UNIVAC in the early 1950's had some form of interrupt vectoring, although it is unclear in exactly which year this feature was available [S08]. Sadly, even in its infancy, we are beginning to lose the origins of computing history.

There is also some debate as to which machine first introduced the idea of DMA. For example, Knuth and others point to the DYSEAC (a "mobile" machine, which at the time meant it could be hauled in a trailer), whereas others think the IBM SAGE may have been the first [S08]. Either way, by the mid 50's, systems with I/O devices that communicated directly with memory and interrupted the CPU when finished existed.

The history here is difficult to trace because the inventions are tied to real, and sometimes obscure, machines. For example, some think that the Lincoln Labs TX-2 machine was first with vectored interrupts [S08], but this is hardly clear.

Because the ideas are relatively obvious — no Einsteinian leap is required to come up with the idea of letting the CPU do something else while a slow I/O is pending — perhaps our focus on "who first?" is misguided. What is certainly clear: as people built these early machines, it became obvious that I/O support was needed. Interrupts, DMA, and related ideas are all direct outcomes of the nature of fast CPUs and slow devices; if you were there at the time, you might have had similar ideas.

36.10 Summary

You should now have a very basic understanding of how an OS interacts with a device. Two techniques, the interrupt and DMA, have been introduced to help with device efficiency, and two approaches to accessing device registers, explicit I/O instructions and memory-mapped I/O, have been described. Finally, the notion of a device driver has been presented, showing how the OS itself can encapsulate low-level details and thus make it easier to build the rest of the OS in a device-neutral fashion.

References

[A+11] "vIC: Interrupt Coalescing for Virtual Machine Storage Device IO"
Irfan Ahmad, Ajay Gulati, Ali Mashtizadeh
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A terrific survey of interrupt coalescing in traditional and virtualized environments.

[C01] "An Empirical Study of Operating System Errors"
Andy Chou, Junfeng Yang, Benjamin Chelf, Seth Hallem, Dawson Engler
SOSP '01
One of the first papers to systematically explore how many bugs are in modern operating systems. Among other neat findings, the authors show that device drivers have something like seven times more bugs than mainline kernel code.

[CK+08] "The xv6 Operating System"
Russ Cox, Frans Kaashoek, Robert Morris, Nickolai Zeldovich
From: http://pdos.csail.mit.edu/6.828/2008/index.html
See `ide.c` for the IDE device driver, with a few more details therein.

[D07] "What Every Programmer Should Know About Memory"
Ulrich Drepper
November, 2007
Available: http://www.akkadia.org/drepper/cpumemory.pdf
A fantastic read about modern memory systems, starting at DRAM and going all the way up to virtualization and cache-optimized algorithms.

[G08] "EIO: Error-handling is Occasionally Correct"
Haryadi Gunawi, Cindy Rubio-Gonzalez, Andrea Arpaci-Dusseau, Remzi Arpaci-Dusseau, Ben Liblit
FAST '08, San Jose, CA, February 2008
Our own work on building a tool to find code in Linux file systems that does not handle error return properly. We found hundreds and hundreds of bugs, many of which have now been fixed.

[L94] "AT Attachment Interface for Disk Drives"
Lawrence J. Lamers, X3T10 Technical Editor
Available: ftp://ftp.t10.org/t13/project/d0791r4c-ATA-1.pdf
Reference number: ANSI X3.221 - 1994 *A rather dry document about device interfaces. Read it at your own peril.*

[MR96] "Eliminating Receive Livelock in an Interrupt-driven Kernel"
Jeffrey Mogul and K. K. Ramakrishnan
USENIX '96, San Diego, CA, January 1996
Mogul and colleagues did a great deal of pioneering work on web server network performance. This paper is but one example.

[S08] "Interrupts"
Mark Smotherman, as of July '08
Available: http://people.cs.clemson.edu/~mark/interrupts.html
A treasure trove of information on the history of interrupts, DMA, and related early ideas in computing.

[S03] "Improving the Reliability of Commodity Operating Systems"
Michael M. Swift, Brian N. Bershad, and Henry M. Levy
SOSP '03
Swift's work revived interest in a more microkernel-like approach to operating systems; minimally, it finally gave some good reasons why address-space based protection could be useful in a modern OS.

[W10] "Hard Disk Driver"
Washington State Course Homepage
Available: http://eecs.wsu.edu/~cs460/cs560/HDdriver.html
A nice summary of a simple IDE disk drive's interface and how to build a device driver for it.

37

Hard Disk Drives

The last chapter introduced the general concept of an I/O device and showed you how the OS might interact with such a beast. In this chapter, we dive into more detail about one device in particular: the **hard disk drive**. These drives have been the main form of persistent data storage in computer systems for decades and much of the development of file system technology (coming soon) is predicated on their behavior. Thus, it is worth understanding the details of a disk's operation before building the file system software that manages it. Many of these details are available in excellent papers by Ruemmler and Wilkes [RW92] and Anderson, Dykes, and Riedel [ADR03].

CRUX: HOW TO STORE AND ACCESS DATA ON DISK
How do modern hard-disk drives store data? What is the interface? How is the data actually laid out and accessed? How does disk scheduling improve performance?

37.1 The Interface

Let's start by understanding the interface to a modern disk drive. The basic interface for all modern drives is straightforward. The drive consists of a large number of sectors (512-byte blocks), each of which can be read or written. The sectors are numbered from 0 to $n - 1$ on a disk with n sectors. Thus, we can view the disk as an array of sectors; 0 to $n - 1$ is thus the **address space** of the drive.

Multi-sector operations are possible; indeed, many file systems will read or write 4KB at a time (or more). However, when updating the disk, the only guarantee drive manufactures make is that a single 512-byte write is **atomic** (i.e., it will either complete in its entirety or it won't complete at all); thus, if an untimely power loss occurs, only a portion of a larger write may complete (sometimes called a **torn write**).

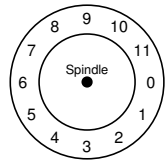

Figure 37.1: **A Disk With Just A Single Track**

There are some assumptions most clients of disk drives make, but that are not specified directly in the interface; Schlosser and Ganger have called this the "unwritten contract" of disk drives [SG04]. Specifically, one can usually assume that accessing two blocks that are near one-another within the drive's address space will be faster than accessing two blocks that are far apart. One can also usually assume that accessing blocks in a contiguous chunk (i.e., a sequential read or write) is the fastest access mode, and usually much faster than any more random access pattern.

37.2 Basic Geometry

Let's start to understand some of the components of a modern disk. We start with a **platter**, a circular hard surface on which data is stored persistently by inducing magnetic changes to it. A disk may have one or more platters; each platter has 2 sides, each of which is called a **surface**. These platters are usually made of some hard material (such as aluminum), and then coated with a thin magnetic layer that enables the drive to persistently store bits even when the drive is powered off.

The platters are all bound together around the **spindle**, which is connected to a motor that spins the platters around (while the drive is powered on) at a constant (fixed) rate. The rate of rotation is often measured in **rotations per minute (RPM)**, and typical modern values are in the 7,200 RPM to 15,000 RPM range. Note that we will often be interested in the time of a single rotation, e.g., a drive that rotates at 10,000 RPM means that a single rotation takes about 6 milliseconds (6 ms).

Data is encoded on each surface in concentric circles of sectors; we call one such concentric circle a **track**. A single surface contains many thousands and thousands of tracks, tightly packed together, with hundreds of tracks fitting into the width of a human hair.

To read and write from the surface, we need a mechanism that allows us to either sense (i.e., read) the magnetic patterns on the disk or to induce a change in (i.e., write) them. This process of reading and writing is accomplished by the **disk head**; there is one such head per surface of the drive. The disk head is attached to a single **disk arm**, which moves across the surface to position the head over the desired track.

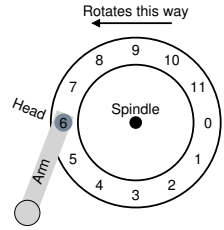

Figure 37.2: **A Single Track Plus A Head**

37.3 A Simple Disk Drive

Let's understand how disks work by building up a model one track at a time. Assume we have a simple disk with a single track (Figure 37.1).

This track has just 12 sectors, each of which is 512 bytes in size (our typical sector size, recall) and addressed therefore by the numbers 0 through 11. The single platter we have here rotates around the spindle, to which a motor is attached. Of course, the track by itself isn't too interesting; we want to be able to read or write those sectors, and thus we need a disk head, attached to a disk arm, as we now see (Figure 37.2).

In the figure, the disk head, attached to the end of the arm, is positioned over sector 6, and the surface is rotating counter-clockwise.

Single-track Latency: The Rotational Delay

To understand how a request would be processed on our simple, one-track disk, imagine we now receive a request to read block 0. How should the disk service this request?

In our simple disk, the disk doesn't have to do much. In particular, it must just wait for the desired sector to rotate under the disk head. This wait happens often enough in modern drives, and is an important enough component of I/O service time, that it has a special name: **rotational delay** (sometimes **rotation delay**, though that sounds weird). In the example, if the full rotational delay is R, the disk has to incur a rotational delay of about $\frac{R}{2}$ to wait for 0 to come under the read/write head (if we start at 6). A worst-case request on this single track would be to sector 5, causing nearly a full rotational delay in order to service such a request.

Multiple Tracks: Seek Time

So far our disk just has a single track, which is not too realistic; modern disks of course have many millions. Let's thus look at ever-so-slightly more realistic disk surface, this one with three tracks (Figure 37.3, left).

In the figure, the head is currently positioned over the innermost track (which contains sectors 24 through 35); the next track over contains the next set of sectors (12 through 23), and the outermost track contains the first sectors (0 through 11).

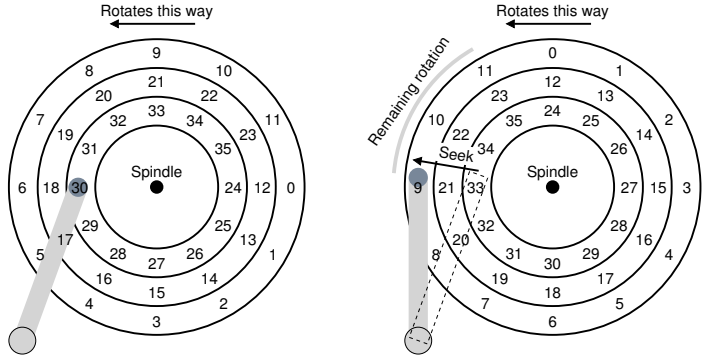

Figure 37.3: **Three Tracks Plus A Head (Right: With Seek)**

To understand how the drive might access a given sector, we now trace what would happen on a request to a distant sector, e.g., a read to sector 11. To service this read, the drive has to first move the disk arm to the correct track (in this case, the outermost one), in a process known as a **seek**. Seeks, along with rotations, are one of the most costly disk operations.

The seek, it should be noted, has many phases: first an *acceleration* phase as the disk arm gets moving; then *coasting* as the arm is moving at full speed, then *deceleration* as the arm slows down; finally *settling* as the head is carefully positioned over the correct track. The **settling time** is often quite significant, e.g., 0.5 to 2 ms, as the drive must be certain to find the right track (imagine if it just got close instead!).

After the seek, the disk arm has positioned the head over the right track. A depiction of the seek is found in Figure 37.3 (right).

As we can see, during the seek, the arm has been moved to the desired track, and the platter of course has rotated, in this case about 3 sectors. Thus, sector 9 is just about to pass under the disk head, and we must only endure a short rotational delay to complete the transfer.

When sector 11 passes under the disk head, the final phase of I/O will take place, known as the **transfer**, where data is either read from or written to the surface. And thus, we have a complete picture of I/O time: first a seek, then waiting for the rotational delay, and finally the transfer.

Some Other Details

Though we won't spend too much time on it, there are some other interesting details about how hard drives operate. Many drives employ some kind of **track skew** to make sure that sequential reads can be properly serviced even when crossing track boundaries. In our simple example disk, this might appear as seen in Figure 37.4.

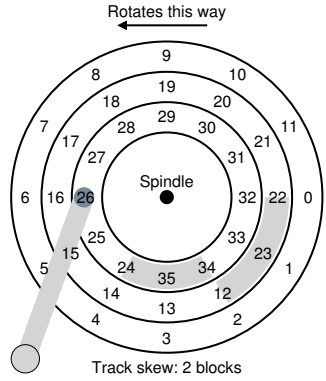

Figure 37.4: **Three Tracks: Track Skew Of 2**

Sectors are often skewed like this because when switching from one track to another, the disk needs time to reposition the head (even to neighboring tracks). Without such skew, the head would be moved to the next track but the desired next block would have already rotated under the head, and thus the drive would have to wait almost the entire rotational delay to access the next block.

Another reality is that outer tracks tend to have more sectors than inner tracks, which is a result of geometry; there is simply more room out there. These tracks are often referred to as **multi-zoned** disk drives, where the disk is organized into multiple zones, and where a zone is consecutive set of tracks on a surface. Each zone has the same number of sectors per track, and outer zones have more sectors than inner zones.

Finally, an important part of any modern disk drive is its **cache**, for historical reasons sometimes called a **track buffer**. This cache is just some small amount of memory (usually around 8 or 16 MB) which the drive can use to hold data read from or written to the disk. For example, when reading a sector from the disk, the drive might decide to read in all of the sectors on that track and cache them in its memory; doing so allows the drive to quickly respond to any subsequent requests to the same track.

On writes, the drive has a choice: should it acknowledge the write has completed when it has put the data in its memory, or after the write has actually been written to disk? The former is called **write back** caching (or sometimes **immediate reporting**), and the latter **write through**. Write back caching sometimes makes the drive appear "faster", but can be dangerous; if the file system or applications require that data be written to disk in a certain order for correctness, write-back caching can lead to problems (read the chapter on file-system journaling for details).

ASIDE: **DIMENSIONAL ANALYSIS**

Remember in Chemistry class, how you solved virtually every problem by simply setting up the units such that they canceled out, and somehow the answers popped out as a result? That chemical magic is known by the highfalutin name of **dimensional analysis** and it turns out it is useful in computer systems analysis too.

Let's do an example to see how dimensional analysis works and why it is useful. In this case, assume you have to figure out how long, in milliseconds, a single rotation of a disk takes. Unfortunately, you are given only the **RPM** of the disk, or **rotations per minute**. Let's assume we're talking about a 10K RPM disk (i.e., it rotates 10,000 times per minute). How do we set up the dimensional analysis so that we get time per rotation in milliseconds?

To do so, we start by putting the desired units on the left; in this case, we wish to obtain the time (in milliseconds) per rotation, so that is exactly what we write down: $\frac{Time\ (ms)}{1\ Rotation}$. We then write down everything we know, making sure to cancel units where possible. First, we obtain $\frac{1\ minute}{10{,}000\ Rotations}$ (keeping rotation on the bottom, as that's where it is on the left), then transform minutes into seconds with $\frac{60\ seconds}{1\ minute}$, and then finally transform seconds in milliseconds with $\frac{1000\ ms}{1\ second}$. The final result is the following (with units nicely canceled):

$$\frac{Time\ (ms)}{1\ Rot.} = \frac{1\ \cancel{minute}}{10{,}000\ Rot.} \cdot \frac{60\ \cancel{seconds}}{1\ \cancel{minute}} \cdot \frac{1000\ ms}{1\ \cancel{second}} = \frac{60{,}000\ ms}{10{,}000\ Rot.} = \frac{6\ ms}{Rotation}$$

As you can see from this example, dimensional analysis makes what seems intuitive into a simple and repeatable process. Beyond the RPM calculation above, it comes in handy with I/O analysis regularly. For example, you will often be given the transfer rate of a disk, e.g., 100 MB/second, and then asked: how long does it take to transfer a 512 KB block (in milliseconds)? With dimensional analysis, it's easy:

$$\frac{Time\ (ms)}{1\ Request} = \frac{512\ \cancel{KB}}{1\ Request} \cdot \frac{1\ \cancel{MB}}{1024\ \cancel{KB}} \cdot \frac{1\ \cancel{second}}{100\ \cancel{MB}} \cdot \frac{1000\ ms}{1\ \cancel{second}} = \frac{5\ ms}{Request}$$

37.4 I/O Time: Doing The Math

Now that we have an abstract model of the disk, we can use a little analysis to better understand disk performance. In particular, we can now represent I/O time as the sum of three major components:

$$T_{I/O} = T_{seek} + T_{rotation} + T_{transfer} \tag{37.1}$$

	Cheetah 15K.5	Barracuda
Capacity	300 GB	1 TB
RPM	15,000	7,200
Average Seek	4 ms	9 ms
Max Transfer	125 MB/s	105 MB/s
Platters	4	4
Cache	16 MB	16/32 MB
Connects via	SCSI	SATA

Figure 37.5: **Disk Drive Specs: SCSI Versus SATA**

Note that the rate of I/O ($R_{I/O}$), which is often more easily used for comparison between drives (as we will do below), is easily computed from the time. Simply divide the size of the transfer by the time it took:

$$R_{I/O} = \frac{Size_{Transfer}}{T_{I/O}} \tag{37.2}$$

To get a better feel for I/O time, let us perform the following calculation. Assume there are two workloads we are interested in. The first, known as the **random** workload, issues small (e.g., 4KB) reads to random locations on the disk. Random workloads are common in many important applications, including database management systems. The second, known as the **sequential** workload, simply reads a large number of sectors consecutively from the disk, without jumping around. Sequential access patterns are quite common and thus important as well.

To understand the difference in performance between random and sequential workloads, we need to make a few assumptions about the disk drive first. Let's look at a couple of modern disks from Seagate. The first, known as the Cheetah 15K.5 [S09b], is a high-performance SCSI drive. The second, the Barracuda [S09a], is a drive built for capacity. Details on both are found in Figure 37.5.

As you can see, the drives have quite different characteristics, and in many ways nicely summarize two important components of the disk drive market. The first is the "high performance" drive market, where drives are engineered to spin as fast as possible, deliver low seek times, and transfer data quickly. The second is the "capacity" market, where cost per byte is the most important aspect; thus, the drives are slower but pack as many bits as possible into the space available.

From these numbers, we can start to calculate how well the drives would do under our two workloads outlined above. Let's start by looking at the random workload. Assuming each 4 KB read occurs at a random location on disk, we can calculate how long each such read would take. On the Cheetah:

$$T_{seek} = 4\ ms,\ T_{rotation} = 2\ ms,\ T_{transfer} = 30\ microsecs \tag{37.3}$$

TIP: USE DISKS SEQUENTIALLY
When at all possible, transfer data to and from disks in a sequential manner. If sequential is not possible, at least think about transferring data in large chunks: the bigger, the better. If I/O is done in little random pieces, I/O performance will suffer dramatically. Also, users will suffer. Also, you will suffer, knowing what suffering you have wrought with your careless random I/Os.

The average seek time (4 milliseconds) is just taken as the average time reported by the manufacturer; note that a full seek (from one end of the surface to the other) would likely take two or three times longer. The average rotational delay is calculated from the RPM directly. 15000 RPM is equal to 250 RPS (rotations per second); thus, each rotation takes 4 ms. On average, the disk will encounter a half rotation and thus 2 ms is the average time. Finally, the transfer time is just the size of the transfer over the peak transfer rate; here it is vanishingly small (30 *microseconds*; note that we need 1000 microseconds just to get 1 millisecond!).

Thus, from our equation above, $T_{I/O}$ for the Cheetah roughly equals 6 ms. To compute the rate of I/O, we just divide the size of the transfer by the average time, and thus arrive at $R_{I/O}$ for the Cheetah under the random workload of about 0.66 MB/s. The same calculation for the Barracuda yields a $T_{I/O}$ of about 13.2 ms, more than twice as slow, and thus a rate of about 0.31 MB/s.

Now let's look at the sequential workload. Here we can assume there is a single seek and rotation before a very long transfer. For simplicity, assume the size of the transfer is 100 MB. Thus, $T_{I/O}$ for the Barracuda and Cheetah is about 800 ms and 950 ms, respectively. The rates of I/O are thus very nearly the peak transfer rates of 125 MB/s and 105 MB/s, respectively. Figure 37.6 summarizes these numbers.

The figure shows us a number of important things. First, and most importantly, there is a huge gap in drive performance between random and sequential workloads, almost a factor of 200 or so for the Cheetah and more than a factor 300 difference for the Barracuda. And thus we arrive at the most obvious design tip in the history of computing.

A second, more subtle point: there is a large difference in performance between high-end "performance" drives and low-end "capacity" drives. For this reason (and others), people are often willing to pay top dollar for the former while trying to get the latter as cheaply as possible.

	Cheetah	Barracuda
$R_{I/O}$ Random	0.66 MB/s	0.31 MB/s
$R_{I/O}$ Sequential	125 MB/s	105 MB/s

Figure 37.6: **Disk Drive Performance: SCSI Versus SATA**

ASIDE: **COMPUTING THE "AVERAGE" SEEK**

In many books and papers, you will see average disk-seek time cited as being roughly one-third of the full seek time. Where does this come from?

Turns out it arises from a simple calculation based on average seek *distance*, not time. Imagine the disk as a set of tracks, from 0 to N. The seek distance between any two tracks x and y is thus computed as the absolute value of the difference between them: $|x - y|$.

To compute the average seek distance, all you need to do is to first add up all possible seek distances:

$$\sum_{x=0}^{N}\sum_{y=0}^{N}|x-y|. \tag{37.4}$$

Then, divide this by the number of different possible seeks: N^2. To compute the sum, we'll just use the integral form:

$$\int_{x=0}^{N}\int_{y=0}^{N}|x-y|\,\mathrm{d}y\,\mathrm{d}x. \tag{37.5}$$

To compute the inner integral, let's break out the absolute value:

$$\int_{y=0}^{x}(x-y)\,\mathrm{d}y+\int_{y=x}^{N}(y-x)\,\mathrm{d}y. \tag{37.6}$$

Solving this leads to $(xy-\frac{1}{2}y^2)\big|_0^x+(\frac{1}{2}y^2-xy)\big|_x^N$ which can be simplified to $(x^2-Nx+\frac{1}{2}N^2)$. Now we have to compute the outer integral:

$$\int_{x=0}^{N}(x^2-Nx+\frac{1}{2}N^2)\,\mathrm{d}x, \tag{37.7}$$

which results in:

$$(\frac{1}{3}x^3-\frac{N}{2}x^2+\frac{N^2}{2}x)\Big|_0^N=\frac{N^3}{3}. \tag{37.8}$$

Remember that we still have to divide by the total number of seeks (N^2) to compute the average seek distance: $(\frac{N^3}{3})/(N^2)=\frac{1}{3}N$. Thus the average seek distance on a disk, over all possible seeks, is one-third the full distance. And now when you hear that an average seek is one-third of a full seek, you'll know where it came from.

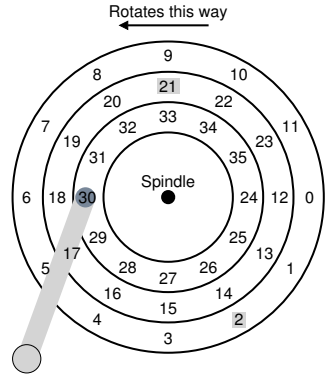

Figure 37.7: **SSTF: Scheduling Requests 21 And 2**

37.5 Disk Scheduling

Because of the high cost of I/O, the OS has historically played a role in deciding the order of I/Os issued to the disk. More specifically, given a set of I/O requests, the **disk scheduler** examines the requests and decides which one to schedule next [SCO90, JW91].

Unlike job scheduling, where the length of each job is usually unknown, with disk scheduling, we can make a good guess at how long a "job" (i.e., disk request) will take. By estimating the seek and possible rotational delay of a request, the disk scheduler can know how long each request will take, and thus (greedily) pick the one that will take the least time to service first. Thus, the disk scheduler will try to follow the **principle of SJF (shortest job first)** in its operation.

SSTF: Shortest Seek Time First

One early disk scheduling approach is known as **shortest-seek-time-first** (**SSTF**) (also called **shortest-seek-first** or **SSF**). SSTF orders the queue of I/O requests by track, picking requests on the nearest track to complete first. For example, assuming the current position of the head is over the inner track, and we have requests for sectors 21 (middle track) and 2 (outer track), we would then issue the request to 21 first, wait for it to complete, and then issue the request to 2 (Figure 37.7).

SSTF works well in this example, seeking to the middle track first and then the outer track. However, SSTF is not a panacea, for the following reasons. First, the drive geometry is not available to the host OS; rather, it sees an array of blocks. Fortunately, this problem is rather easily fixed. Instead of SSTF, an OS can simply implement **nearest-block-first** (**NBF**), which schedules the request with the nearest block address next.

The second problem is more fundamental: **starvation**. Imagine in our example above if there were a steady stream of requests to the inner track, where the head currently is positioned. Requests to any other tracks would then be ignored completely by a pure SSTF approach. And thus the crux of the problem:

CRUX: HOW TO HANDLE DISK STARVATION
How can we implement SSTF-like scheduling but avoid starvation?

Elevator (a.k.a. SCAN or C-SCAN)

The answer to this query was developed some time ago (see [CKR72] for example), and is relatively straightforward. The algorithm, originally called **SCAN**, simply moves back and forth across the disk servicing requests in order across the tracks. Let's call a single pass across the disk (from outer to inner tracks, or inner to outer) a *sweep*. Thus, if a request comes for a block on a track that has already been serviced on this sweep of the disk, it is not handled immediately, but rather queued until the next sweep (in the other direction).

SCAN has a number of variants, all of which do about the same thing. For example, Coffman et al. introduced **F-SCAN**, which freezes the queue to be serviced when it is doing a sweep [CKR72]; this action places requests that come in during the sweep into a queue to be serviced later. Doing so avoids starvation of far-away requests, by delaying the servicing of late-arriving (but nearer by) requests.

C-SCAN is another common variant, short for **Circular SCAN**. Instead of sweeping in both directions across the disk, the algorithm only sweeps from outer-to-inner, and then resets at the outer track to begin again. Doing so is a bit more fair to inner and outer tracks, as pure back-and-forth SCAN favors the middle tracks, i.e., after servicing the outer track, SCAN passes through the middle twice before coming back to the outer track again.

For reasons that should now be clear, the SCAN algorithm (and its cousins) is sometimes referred to as the **elevator** algorithm, because it behaves like an elevator which is either going up or down and not just servicing requests to floors based on which floor is closer. Imagine how annoying it would be if you were going down from floor 10 to 1, and somebody got on at 3 and pressed 4, and the elevator went up to 4 because it was "closer" than 1! As you can see, the elevator algorithm, when used in real life, prevents fights from taking place on elevators. In disks, it just prevents starvation.

Unfortunately, SCAN and its cousins do not represent the best scheduling technology. In particular, SCAN (or SSTF even) do not actually adhere as closely to the principle of SJF as they could. In particular, they ignore rotation. And thus, another crux:

CRUX: HOW TO ACCOUNT FOR DISK ROTATION COSTS
How can we implement an algorithm that more closely approximates SJF by taking *both* seek and rotation into account?

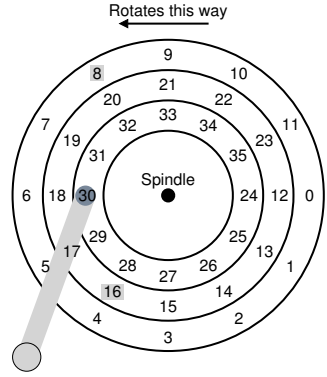

Figure 37.8: **SSTF: Sometimes Not Good Enough**

SPTF: Shortest Positioning Time First

Before discussing **shortest positioning time first** or **SPTF** scheduling (sometimes also called **shortest access time first** or **SATF**), which is the solution to our problem, let us make sure we understand the problem in more detail. Figure 37.8 presents an example.

In the example, the head is currently positioned over sector 30 on the inner track. The scheduler thus has to decide: should it schedule sector 16 (on the middle track) or sector 8 (on the outer track) for its next request. So which should it service next?

The answer, of course, is "it depends". In engineering, it turns out "it depends" is almost always the answer, reflecting that trade-offs are part of the life of the engineer; such maxims are also good in a pinch, e.g., when you don't know an answer to your boss's question, you might want to try this gem. However, it is almost always better to know *why* it depends, which is what we discuss here.

What it depends on here is the relative time of seeking as compared to rotation. If, in our example, seek time is much higher than rotational delay, then SSTF (and variants) are just fine. However, imagine if seek is quite a bit faster than rotation. Then, in our example, it would make more sense to seek *further* to service request 8 on the outer track than it would to perform the shorter seek to the middle track to service 16, which has to rotate all the way around before passing under the disk head.

On modern drives, as we saw above, both seek and rotation are roughly

TIP: IT ALWAYS DEPENDS (LIVNY'S LAW)
Almost any question can be answered with "it depends", as our colleague Miron Livny always says. However, use with caution, as if you answer too many questions this way, people will stop asking you questions altogether. For example, somebody asks: "want to go to lunch?" You reply: "it depends, are *you* coming along?"

equivalent (depending, of course, on the exact requests), and thus SPTF is useful and improves performance. However, it is even more difficult to implement in an OS, which generally does not have a good idea where track boundaries are or where the disk head currently is (in a rotational sense). Thus, SPTF is usually performed inside a drive, described below.

Other Scheduling Issues

There are many other issues we do not discuss in this brief description of basic disk operation, scheduling, and related topics. One such issue is this: *where* is disk scheduling performed on modern systems? In older systems, the operating system did all the scheduling; after looking through the set of pending requests, the OS would pick the best one, and issue it to the disk. When that request completed, the next one would be chosen, and so forth. Disks were simpler then, and so was life.

In modern systems, disks can accommodate multiple outstanding requests, and have sophisticated internal schedulers themselves (which can implement SPTF accurately; inside the disk controller, all relevant details are available, including exact head position). Thus, the OS scheduler usually picks what it thinks the best few requests are (say 16) and issues them all to disk; the disk then uses its internal knowledge of head position and detailed track layout information to service said requests in the best possible (SPTF) order.

Another important related task performed by disk schedulers is **I/O merging**. For example, imagine a series of requests to read blocks 33, then 8, then 34, as in Figure 37.8. In this case, the scheduler should **merge** the requests for blocks 33 and 34 into a single two-block request; any reordering that the scheduler does is performed upon the merged requests. Merging is particularly important at the OS level, as it reduces the number of requests sent to the disk and thus lowers overheads.

One final problem that modern schedulers address is this: how long should the system wait before issuing an I/O to disk? One might naively think that the disk, once it has even a single I/O, should immediately issue the request to the drive; this approach is called **work-conserving**, as the disk will never be idle if there are requests to serve. However, research on **anticipatory disk scheduling** has shown that sometimes it is better to wait for a bit [ID01], in what is called a **non-work-conserving** approach.

By waiting, a new and "better" request may arrive at the disk, and thus overall efficiency is increased. Of course, deciding when to wait, and for how long, can be tricky; see the research paper for details, or check out the Linux kernel implementation to see how such ideas are transitioned into practice (if you are the ambitious sort).

37.6 Summary

We have presented a summary of how disks work. The summary is actually a detailed functional model; it does not describe the amazing physics, electronics, and material science that goes into actual drive design. For those interested in even more details of that nature, we suggest a different major (or perhaps minor); for those that are happy with this model, good! We can now proceed to using the model to build more interesting systems on top of these incredible devices.

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[SCO90] "Disk Scheduling Revisited"
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USENIX 1990
A paper that talks about how rotation matters too in the world of disk scheduling.

[SG04] "MEMS-based storage devices and standard disk interfaces:
A square peg in a round hole?"
Steven W. Schlosser, Gregory R. Ganger
FAST '04, pp. 87-100, 2004
While the MEMS aspect of this paper hasn't yet made an impact, the discussion of the contract between file systems and disks is wonderful and a lasting contribution.

[S09a] "Barracuda ES.2 data sheet"
`http://www.seagate.com/docs/pdf/datasheet/disc/ds_barracuda_es.pdf`
A data sheet; read at your own risk. Risk of what? Boredom.

[S09b] "Cheetah 15K.5"
`http://www.seagate.com/docs/pdf/datasheet/disc/ds-cheetah-15k-5-us.pdf`
See above commentary on data sheets.

Homework

This homework uses `disk.py` to familiarize you with how a modern hard drive works. It has a lot of different options, and unlike most of the other simulations, has a graphical animator to show you exactly what happens when the disk is in action. See the README for details.

1. Compute the seek, rotation, and transfer times for the following sets of requests: `-a 0`, `-a 6`, `-a 30`, `-a 7,30,8`, and finally `-a 10,11,12,13`.
2. Do the same requests above, but change the seek rate to different values: `-S 2`, `-S 4`, `-S 8`, `-S 10`, `-S 40`, `-S 0.1`. How do the times change?
3. Do the same requests above, but change the rotation rate: `-R 0.1`, `-R 0.5`, `-R 0.01`. How do the times change?
4. You might have noticed that some request streams would be better served with a policy better than FIFO. For example, with the request stream `-a 7,30,8`, what order should the requests be processed in? Now run the shortest seek-time first (SSTF) scheduler (`-p SSTF`) on the same workload; how long should it take (seek, rotation, transfer) for each request to be served?
5. Now do the same thing, but using the shortest access-time first (SATF) scheduler (`-p SATF`). Does it make any difference for the set of requests as specified by `-a 7,30,8`? Find a set of requests where SATF does noticeably better than SSTF; what are the conditions for a noticeable difference to arise?
6. You might have noticed that the request stream `-a 10,11,12,13` wasn't particularly well handled by the disk. Why is that? Can you introduce a track skew to address this problem (`-o skew`, where `skew` is a non-negative integer)? Given the default seek rate, what should the skew be to minimize the total time for this set of requests? What about for different seek rates (e.g., `-S 2`, `-S 4`)? In general, could you write a formula to figure out the skew, given the seek rate and sector layout information?
7. Multi-zone disks pack more sectors into the outer tracks. To configure this disk in such a way, run with the `-z` flag. Specifically, try running some requests against a disk run with `-z 10,20,30` (the numbers specify the angular space occupied by a sector, per track; in this example, the outer track will be packed with a sector every 10 degrees, the middle track every 20 degrees, and the inner track with a sector every 30 degrees). Run some random requests (e.g., `-a -1 -A 5,-1,0`, which specifies that random requests should be used via the `-a -1` flag and that five requests ranging from 0 to the max be generated), and see if you can compute the seek, rotation, and transfer times. Use different random seeds (`-s 1`, `-s 2`, etc.). What is the bandwidth (in sectors per unit time) on the outer, middle, and inner tracks?

8. Scheduling windows determine how many sector requests a disk can examine at once in order to determine which sector to serve next. Generate some random workloads of a lot of requests (e.g., `-A 1000,-1,0`, with different seeds perhaps) and see how long the SATF scheduler takes when the scheduling window is changed from 1 up to the number of requests (e.g., `-w 1` up to `-w 1000`, and some values in between). How big of scheduling window is needed to approach the best possible performance? Make a graph and see. Hint: use the `-c` flag and don't turn on graphics with `-G` to run these more quickly. When the scheduling window is set to 1, does it matter which policy you are using?
9. Avoiding starvation is important in a scheduler. Can you think of a series of requests such that a particular sector is delayed for a very long time given a policy such as SATF? Given that sequence, how does it perform if you use a **bounded SATF** or **BSATF** scheduling approach? In this approach, you specify the scheduling window (e.g., `-w 4`) as well as the BSATF policy (`-p BSATF`); the scheduler then will only move onto the next window of requests when *all* of the requests in the current window have been serviced. Does this solve the starvation problem? How does it perform, as compared to SATF? In general, how should a disk make this trade-off between performance and starvation avoidance?
10. All the scheduling policies we have looked at thus far are **greedy**, in that they simply pick the next best option instead of looking for the optimal schedule over a set of requests. Can you find a set of requests in which this greedy approach is not optimal?

38

Redundant Arrays of Inexpensive Disks (RAIDs)

When we use a disk, we sometimes wish it to be faster; I/O operations are slow and thus can be the bottleneck for the entire system. When we use a disk, we sometimes wish it to be larger; more and more data is being put online and thus our disks are getting fuller and fuller. When we use a disk, we sometimes wish for it to be more reliable; when a disk fails, if our data isn't backed up, all that valuable data is gone.

CRUX: HOW TO MAKE A LARGE, FAST, RELIABLE DISK
How can we make a large, fast, and reliable storage system? What are the key techniques? What are trade-offs between different approaches?

In this chapter, we introduce the **Redundant Array of Inexpensive Disks** better known as **RAID** [P+88], a technique to use multiple disks in concert to build a faster, bigger, and more reliable disk system. The term was introduced in the late 1980s by a group of researchers at U.C. Berkeley (led by Professors David Patterson and Randy Katz and then student Garth Gibson); it was around this time that many different researchers simultaneously arrived upon the basic idea of using multiple disks to build a better storage system [BG88, K86,K88,PB86,SG86].

Externally, a RAID looks like a disk: a group of blocks one can read or write. Internally, the RAID is a complex beast, consisting of multiple disks, memory (both volatile and non-), and one or more processors to manage the system. A hardware RAID is very much like a computer system, specialized for the task of managing a group of disks.

RAIDs offer a number of advantages over a single disk. One advantage is *performance*. Using multiple disks in parallel can greatly speed up I/O times. Another benefit is *capacity*. Large data sets demand large disks. Finally, RAIDs can improve *reliability*; spreading data across multiple disks (without RAID techniques) makes the data vulnerable to the loss of a single disk; with some form of **redundancy**, RAIDs can tolerate the loss of a disk and keep operating as if nothing were wrong.

TIP: TRANSPARENCY ENABLES DEPLOYMENT
When considering how to add new functionality to a system, one should always consider whether such functionality can be added **transparently**, in a way that demands no changes to the rest of the system. Requiring a complete rewrite of the existing software (or radical hardware changes) lessens the chance of impact of an idea. RAID is a perfect example, and certainly its transparency contributed to its success; administrators could install a SCSI-based RAID storage array instead of a SCSI disk, and the rest of the system (host computer, OS, etc.) did not have to change one bit to start using it. By solving this problem of **deployment**, RAID was made more successful from day one.

Amazingly, RAIDs provide these advantages **transparently** to systems that use them, i.e., a RAID just looks like a big disk to the host system. The beauty of transparency, of course, is that it enables one to simply replace a disk with a RAID and not change a single line of software; the operating system and client applications continue to operate without modification. In this manner, transparency greatly improves the **deployability** of RAID, enabling users and administrators to put a RAID to use without worries of software compatibility.

We now discuss some of the important aspects of RAIDs. We begin with the interface, fault model, and then discuss how one can evaluate a RAID design along three important axes: capacity, reliability, and performance. We then discuss a number of other issues that are important to RAID design and implementation.

38.1 Interface And RAID Internals

To a file system above, a RAID looks like a big, (hopefully) fast, and (hopefully) reliable disk. Just as with a single disk, it presents itself as a linear array of blocks, each of which can be read or written by the file system (or other client).

When a file system issues a *logical I/O* request to the RAID, the RAID internally must calculate which disk (or disks) to access in order to complete the request, and then issue one or more *physical I/Os* to do so. The exact nature of these physical I/Os depends on the RAID level, as we will discuss in detail below. However, as a simple example, consider a RAID that keeps two copies of each block (each one on a separate disk); when writing to such a **mirrored** RAID system, the RAID will have to perform two physical I/Os for every one logical I/O it is issued.

A RAID system is often built as a separate hardware box, with a standard connection (e.g., SCSI, or SATA) to a host. Internally, however, RAIDs are fairly complex, consisting of a microcontroller that runs firmware to direct the operation of the RAID, volatile memory such as DRAM to buffer data blocks as they are read and written, and in some cases,

non-volatile memory to buffer writes safely and perhaps even specialized logic to perform parity calculations (useful in some RAID levels, as we will also see below). At a high level, a RAID is very much a specialized computer system: it has a processor, memory, and disks; however, instead of running applications, it runs specialized software designed to operate the RAID.

38.2 Fault Model

To understand RAID and compare different approaches, we must have a fault model in mind. RAIDs are designed to detect and recover from certain kinds of disk faults; thus, knowing exactly which faults to expect is critical in arriving upon a working design.

The first fault model we will assume is quite simple, and has been called the **fail-stop** fault model [S84]. In this model, a disk can be in exactly one of two states: working or failed. With a working disk, all blocks can be read or written. In contrast, when a disk has failed, we assume it is permanently lost.

One critical aspect of the fail-stop model is what it assumes about fault detection. Specifically, when a disk has failed, we assume that this is easily detected. For example, in a RAID array, we would assume that the RAID controller hardware (or software) can immediately observe when a disk has failed.

Thus, for now, we do not have to worry about more complex "silent" failures such as disk corruption. We also do not have to worry about a single block becoming inaccessible upon an otherwise working disk (sometimes called a latent sector error). We will consider these more complex (and unfortunately, more realistic) disk faults later.

38.3 How To Evaluate A RAID

As we will soon see, there are a number of different approaches to building a RAID. Each of these approaches has different characteristics which are worth evaluating, in order to understand their strengths and weaknesses.

Specifically, we will evaluate each RAID design along three axes. The first axis is **capacity**; given a set of N disks each with B blocks, how much useful capacity is available to clients of the RAID? Without redundancy, the answer is $N \cdot B$; in contrast, if we have a system that keeps two copies of each block (called **mirroring**), we obtain a useful capacity of $(N \cdot B)/2$. Different schemes (e.g., parity-based ones) tend to fall in between.

The second axis of evaluation is **reliability**. How many disk faults can the given design tolerate? In alignment with our fault model, we assume only that an entire disk can fail; in later chapters (i.e., on data integrity), we'll think about how to handle more complex failure modes.

Finally, the third axis is **performance**. Performance is somewhat chal-

lenging to evaluate, because it depends heavily on the workload presented to the disk array. Thus, before evaluating performance, we will first present a set of typical workloads that one should consider.

We now consider three important RAID designs: RAID Level 0 (striping), RAID Level 1 (mirroring), and RAID Levels 4/5 (parity-based redundancy). The naming of each of these designs as a "level" stems from the pioneering work of Patterson, Gibson, and Katz at Berkeley [P+88].

38.4 RAID Level 0: Striping

The first RAID level is actually not a RAID level at all, in that there is no redundancy. However, RAID level 0, or **striping** as it is better known, serves as an excellent upper-bound on performance and capacity and thus is worth understanding.

The simplest form of striping will **stripe** blocks across the disks of the system as follows (assume here a 4-disk array):

Disk 0	Disk 1	Disk 2	Disk 3
0	1	2	3
4	5	6	7
8	9	10	11
12	13	14	15

Figure 38.1: **RAID-0: Simple Striping**

From Figure 38.1, you get the basic idea: spread the blocks of the array across the disks in a round-robin fashion. This approach is designed to extract the most parallelism from the array when requests are made for contiguous chunks of the array (as in a large, sequential read, for example). We call the blocks in the same row a **stripe**; thus, blocks 0, 1, 2, and 3 are in the same stripe above.

In the example, we have made the simplifying assumption that only 1 block (each of say size 4KB) is placed on each disk before moving on to the next. However, this arrangement need not be the case. For example, we could arrange the blocks across disks as in Figure 38.2:

Disk 0	Disk 1	Disk 2	Disk 3	
0	2	4	6	chunk size:
1	3	5	7	2 blocks
8	10	12	14	
9	11	13	15	

Figure 38.2: **Striping with a Bigger Chunk Size**

In this example, we place two 4KB blocks on each disk before moving on to the next disk. Thus, the **chunk size** of this RAID array is 8KB, and a stripe thus consists of 4 chunks or 32KB of data.

ASIDE: **THE RAID MAPPING PROBLEM**

Before studying the capacity, reliability, and performance characteristics of the RAID, we first present an aside on what we call **the mapping problem**. This problem arises in all RAID arrays; simply put, given a logical block to read or write, how does the RAID know exactly which physical disk and offset to access?

For these simple RAID levels, we do not need much sophistication in order to correctly map logical blocks onto their physical locations. Take the first striping example above (chunk size = 1 block = 4KB). In this case, given a logical block address A, the RAID can easily compute the desired disk and offset with two simple equations:

```
Disk   = A % number_of_disks
Offset = A / number_of_disks
```

Note that these are all integer operations (e.g., 4 / 3 = 1 not 1.33333...).

Let's see how these equations work for a simple example. Imagine in the first RAID above that a request arrives for block 14. Given that there are 4 disks, this would mean that the disk we are interested in is (14 % 4 = 2): disk 2. The exact block is calculated as (14 / 4 = 3): block 3. Thus, block 14 should be found on the fourth block (block 3, starting at 0) of the third disk (disk 2, starting at 0), which is exactly where it is.

You can think about how these equations would be modified to support different chunk sizes. Try it! It's not too hard.

Chunk Sizes

Chunk size mostly affects performance of the array. For example, a small chunk size implies that many files will get striped across many disks, thus increasing the parallelism of reads and writes to a single file; however, the positioning time to access blocks across multiple disks increases, because the positioning time for the entire request is determined by the maximum of the positioning times of the requests across all drives.

A big chunk size, on the other hand, reduces such intra-file parallelism, and thus relies on multiple concurrent requests to achieve high throughput. However, large chunk sizes reduce positioning time; if, for example, a single file fits within a chunk and thus is placed on a single disk, the positioning time incurred while accessing it will just be the positioning time of a single disk.

Thus, determining the "best" chunk size is hard to do, as it requires a great deal of knowledge about the workload presented to the disk system [CL95]. For the rest of this discussion, we will assume that the array uses a chunk size of a single block (4KB). Most arrays use larger chunk sizes (e.g., 64 KB), but for the issues we discuss below, the exact chunk size does not matter; thus we use a single block for the sake of simplicity.

Back To RAID-0 Analysis

Let us now evaluate the capacity, reliability, and performance of striping. From the perspective of capacity, it is perfect: given N disks each of size B blocks, striping delivers $N \cdot B$ blocks of useful capacity. From the standpoint of reliability, striping is also perfect, but in the bad way: any disk failure will lead to data loss. Finally, performance is excellent: all disks are utilized, often in parallel, to service user I/O requests.

Evaluating RAID Performance

In analyzing RAID performance, one can consider two different performance metrics. The first is *single-request latency*. Understanding the latency of a single I/O request to a RAID is useful as it reveals how much parallelism can exist during a single logical I/O operation. The second is *steady-state throughput* of the RAID, i.e., the total bandwidth of many concurrent requests. Because RAIDs are often used in high-performance environments, the steady-state bandwidth is critical, and thus will be the main focus of our analyses.

To understand throughput in more detail, we need to put forth some workloads of interest. We will assume, for this discussion, that there are two types of workloads: **sequential** and **random**. With a sequential workload, we assume that requests to the array come in large contiguous chunks; for example, a request (or series of requests) that accesses 1 MB of data, starting at block x and ending at block (x+1 MB), would be deemed sequential. Sequential workloads are common in many environments (think of searching through a large file for a keyword), and thus are considered important.

For random workloads, we assume that each request is rather small, and that each request is to a different random location on disk. For example, a random stream of requests may first access 4KB at logical address 10, then at logical address 550,000, then at 20,100, and so forth. Some important workloads, such as transactional workloads on a database management system (DBMS), exhibit this type of access pattern, and thus it is considered an important workload.

Of course, real workloads are not so simple, and often have a mix of sequential and random-seeming components as well as behaviors in-between the two. For simplicity, we just consider these two possibilities.

As you can tell, sequential and random workloads will result in widely different performance characteristics from a disk. With sequential access, a disk operates in its most efficient mode, spending little time seeking and waiting for rotation and most of its time transferring data. With random access, just the opposite is true: most time is spent seeking and waiting for rotation and relatively little time is spent transferring data. To capture this difference in our analysis, we will assume that a disk can transfer data at S MB/s under a sequential workload, and R MB/s when under a random workload. In general, S is much greater than R (i.e., $S \gg R$).

To make sure we understand this difference, let's do a simple exercise. Specifically, let's calculate S and R given the following disk characteristics. Assume a sequential transfer of size 10 MB on average, and a random transfer of 10 KB on average. Also, assume the following disk characteristics:

Average seek time	7 ms
Average rotational delay	3 ms
Transfer rate of disk	50 MB/s

To compute S, we need to first figure out how time is spent in a typical 10 MB transfer. First, we spend 7 ms seeking, and then 3 ms rotating. Finally, transfer begins; 10 MB @ 50 MB/s leads to 1/5th of a second, or 200 ms, spent in transfer. Thus, for each 10 MB request, we spend 210 ms completing the request. To compute S, we just need to divide:

$$S = \frac{Amount\ of\ Data}{Time\ to\ access} = \frac{10\ MB}{210\ ms} = 47.62\ MB/s$$

As we can see, because of the large time spent transferring data, S is very near the peak bandwidth of the disk (the seek and rotational costs have been amortized).

We can compute R similarly. Seek and rotation are the same; we then compute the time spent in transfer, which is 10 KB @ 50 MB/s, or 0.195 ms.

$$R = \frac{Amount\ of\ Data}{Time\ to\ access} = \frac{10\ KB}{10.195\ ms} = 0.981\ MB/s$$

As we can see, R is less than 1 MB/s, and S/R is almost 50.

Back To RAID-0 Analysis, Again

Let's now evaluate the performance of striping. As we said above, it is generally good. From a latency perspective, for example, the latency of a single-block request should be just about identical to that of a single disk; after all, RAID-0 will simply redirect that request to one of its disks.

From the perspective of steady-state throughput, we'd expect to get the full bandwidth of the system. Thus, throughput equals N (the number of disks) multiplied by S (the sequential bandwidth of a single disk). For a large number of random I/Os, we can again use all of the disks, and thus obtain $N \cdot R$ MB/s. As we will see below, these values are both the simplest to calculate and will serve as an upper bound in comparison with other RAID levels.

38.5 RAID Level 1: Mirroring

Our first RAID level beyond striping is known as RAID level 1, or mirroring. With a mirrored system, we simply make more than one copy of each block in the system; each copy should be placed on a separate disk, of course. By doing so, we can tolerate disk failures.

In a typical mirrored system, we will assume that for each logical block, the RAID keeps two physical copies of it. Here is an example:

Disk 0	Disk 1	Disk 2	Disk 3
0	0	1	1
2	2	3	3
4	4	5	5
6	6	7	7

Figure 38.3: **Simple RAID-1: Mirroring**

In the example, disk 0 and disk 1 have identical contents, and disk 2 and disk 3 do as well; the data is striped across these mirror pairs. In fact, you may have noticed that there are a number of different ways to place block copies across the disks. The arrangement above is a common one and is sometimes called **RAID-10** or (**RAID 1+0**) because it uses mirrored pairs (RAID-1) and then stripes (RAID-0) on top of them; another common arrangement is **RAID-01** (or **RAID 0+1**), which contains two large striping (RAID-0) arrays, and then mirrors (RAID-1) on top of them. For now, we will just talk about mirroring assuming the above layout.

When reading a block from a mirrored array, the RAID has a choice: it can read either copy. For example, if a read to logical block 5 is issued to the RAID, it is free to read it from either disk 2 or disk 3. When writing a block, though, no such choice exists: the RAID must update *both* copies of the data, in order to preserve reliability. Do note, though, that these writes can take place in parallel; for example, a write to logical block 5 could proceed to disks 2 and 3 at the same time.

RAID-1 Analysis

Let us assess RAID-1. From a capacity standpoint, RAID-1 is expensive; with the mirroring level = 2, we only obtain half of our peak useful capacity. With N disks of B blocks, RAID-1 useful capacity is $(N \cdot B)/2$.

From a reliability standpoint, RAID-1 does well. It can tolerate the failure of any one disk. You may also notice RAID-1 can actually do better than this, with a little luck. Imagine, in the figure above, that disk 0 and disk 2 both failed. In such a situation, there is no data loss! More generally, a mirrored system (with mirroring level of 2) can tolerate 1 disk failure for certain, and up to N/2 failures depending on which disks fail. In practice, we generally don't like to leave things like this to chance; thus most people consider mirroring to be good for handling a single failure.

Finally, we analyze performance. From the perspective of the latency of a single read request, we can see it is the same as the latency on a single disk; all the RAID-1 does is direct the read to one of its copies. A write is a little different: it requires two physical writes to complete before it is done. These two writes happen in parallel, and thus the time will be roughly equivalent to the time of a single write; however, because the logical write must wait for both physical writes to complete, it suffers the worst-case seek and rotational delay of the two requests, and thus (on average) will be slightly higher than a write to a single disk.

ASIDE: **THE RAID CONSISTENT-UPDATE PROBLEM**

Before analyzing RAID-1, let us first discuss a problem that arises in any multi-disk RAID system, known as the **consistent-update problem** [DAA05]. The problem occurs on a write to any RAID that has to update multiple disks during a single logical operation. In this case, let us assume we are considering a mirrored disk array.

Imagine the write is issued to the RAID, and then the RAID decides that it must be written to two disks, disk 0 and disk 1. The RAID then issues the write to disk 0, but just before the RAID can issue the request to disk 1, a power loss (or system crash) occurs. In this unfortunate case, let us assume that the request to disk 0 completed (but clearly the request to disk 1 did not, as it was never issued).

The result of this untimely power loss is that the two copies of the block are now **inconsistent**; the copy on disk 0 is the new version, and the copy on disk 1 is the old. What we would like to happen is for the state of both disks to change **atomically**, i.e., either both should end up as the new version or neither.

The general way to solve this problem is to use a **write-ahead log** of some kind to first record what the RAID is about to do (i.e., update two disks with a certain piece of data) before doing it. By taking this approach, we can ensure that in the presence of a crash, the right thing will happen; by running a **recovery** procedure that replays all pending transactions to the RAID, we can ensure that no two mirrored copies (in the RAID-1 case) are out of sync.

One last note: because logging to disk on every write is prohibitively expensive, most RAID hardware includes a small amount of non-volatile RAM (e.g., battery-backed) where it performs this type of logging. Thus, consistent update is provided without the high cost of logging to disk.

To analyze steady-state throughput, let us start with the sequential workload. When writing out to disk sequentially, each logical write must result in two physical writes; for example, when we write logical block 0 (in the figure above), the RAID internally would write it to both disk 0 and disk 1. Thus, we can conclude that the maximum bandwidth obtained during sequential writing to a mirrored array is $(\frac{N}{2} \cdot S)$, or half the peak bandwidth.

Unfortunately, we obtain the exact same performance during a sequential read. One might think that a sequential read could do better, because it only needs to read one copy of the data, not both. However, let's use an example to illustrate why this doesn't help much. Imagine we need to read blocks 0, 1, 2, 3, 4, 5, 6, and 7. Let's say we issue the read of 0 to disk 0, the read of 1 to disk 2, the read of 2 to disk 1, and the read of 3 to disk 3. We continue by issuing reads to 4, 5, 6, and 7 to disks 0, 2, 1, and 3, respectively. One might naively think that because we are utilizing all disks, we are achieving the full bandwidth of the array.

To see that this is not (necessarily) the case, however, consider the

requests a single disk receives (say disk 0). First, it gets a request for block 0; then, it gets a request for block 4 (skipping block 2). In fact, each disk receives a request for every other block. While it is rotating over the skipped block, it is not delivering useful bandwidth to the client. Thus, each disk will only deliver half its peak bandwidth. And thus, the sequential read will only obtain a bandwidth of ($\frac{N}{2} \cdot S$) MB/s.

Random reads are the best case for a mirrored RAID. In this case, we can distribute the reads across all the disks, and thus obtain the full possible bandwidth. Thus, for random reads, RAID-1 delivers $N \cdot R$ MB/s.

Finally, random writes perform as you might expect: $\frac{N}{2} \cdot R$ MB/s. Each logical write must turn into two physical writes, and thus while all the disks will be in use, the client will only perceive this as half the available bandwidth. Even though a write to logical block x turns into two parallel writes to two different physical disks, the bandwidth of many small requests only achieves half of what we saw with striping. As we will soon see, getting half the available bandwidth is actually pretty good!

38.6 RAID Level 4: Saving Space With Parity

We now present a different method of adding redundancy to a disk array known as **parity**. Parity-based approaches attempt to use less capacity and thus overcome the huge space penalty paid by mirrored systems. They do so at a cost, however: performance.

Disk 0	Disk 1	Disk 2	Disk 3	Disk 4
0	1	2	3	P0
4	5	6	7	P1
8	9	10	11	P2
12	13	14	15	P3

Figure 38.4: **RAID-4 with Parity**

Here is an example five-disk RAID-4 system (Figure 38.4). For each stripe of data, we have added a single **parity** block that stores the redundant information for that stripe of blocks. For example, parity block P1 has redundant information that it calculated from blocks 4, 5, 6, and 7.

To compute parity, we need to use a mathematical function that enables us to withstand the loss of any one block from our stripe. It turns out the simple function **XOR** does the trick quite nicely. For a given set of bits, the XOR of all of those bits returns a 0 if there are an even number of 1's in the bits, and a 1 if there are an odd number of 1's. For example:

C0	C1	C2	C3	P
0	0	1	1	XOR(0,0,1,1) = 0
0	1	0	0	XOR(0,1,0,0) = 1

In the first row (0,0,1,1), there are two 1's (C2, C3), and thus XOR of all of those values will be 0 (P); similarly, in the second row there is only one 1 (C1), and thus the XOR must be 1 (P). You can remember this in a simple way: that the number of 1s in any row must be an even (not odd) number; that is the **invariant** that the RAID must maintain in order for parity to be correct.

From the example above, you might also be able to guess how parity information can be used to recover from a failure. Imagine the column labeled C2 is lost. To figure out what values must have been in the column, we simply have to read in all the other values in that row (including the XOR'd parity bit) and **reconstruct** the right answer. Specifically, assume the first row's value in column C2 is lost (it is a 1); by reading the other values in that row (0 from C0, 0 from C1, 1 from C3, and 0 from the parity column P), we get the values 0, 0, 1, and 0. Because we know that XOR keeps an even number of 1's in each row, we know what the missing data must be: a 1. And that is how reconstruction works in a XOR-based parity scheme! Note also how we compute the reconstructed value: we just XOR the data bits and the parity bits together, in the same way that we calculated the parity in the first place.

Now you might be wondering: we are talking about XORing all of these bits, and yet from above we know that the RAID places 4KB (or larger) blocks on each disk; how do we apply XOR to a bunch of blocks to compute the parity? It turns out this is easy as well. Simply perform a bitwise XOR across each bit of the data blocks; put the result of each bitwise XOR into the corresponding bit slot in the parity block. For example, if we had blocks of size 4 bits (yes, this is still quite a bit smaller than a 4KB block, but you get the picture), they might look something like this:

Block0	Block1	Block2	Block3	Parity
00	10	11	10	11
10	01	00	01	10

As you can see from the figure, the parity is computed for each bit of each block and the result placed in the parity block.

RAID-4 Analysis

Let us now analyze RAID-4. From a capacity standpoint, RAID-4 uses 1 disk for parity information for every group of disks it is protecting. Thus, our useful capacity for a RAID group is $(N - 1) \cdot B$.

Reliability is also quite easy to understand: RAID-4 tolerates 1 disk failure and no more. If more than one disk is lost, there is simply no way to reconstruct the lost data.

Finally, there is performance. This time, let us start by analyzing steady-state throughput. Sequential read performance can utilize all of the disks except for the parity disk, and thus deliver a peak effective bandwidth of $(N - 1) \cdot S$ MB/s (an easy case).

Disk 0	Disk 1	Disk 2	Disk 3	Disk 4
0	1	2	3	P0
4	5	6	7	P1
8	9	10	11	P2
12	13	14	15	P3

Figure 38.5: **Full-stripe Writes In RAID-4**

To understand the performance of sequential writes, we must first understand how they are done. When writing a big chunk of data to disk, RAID-4 can perform a simple optimization known as a **full-stripe write**. For example, imagine the case where the blocks 0, 1, 2, and 3 have been sent to the RAID as part of a write request (Figure 38.5).

In this case, the RAID can simply calculate the new value of P0 (by performing an XOR across the blocks 0, 1, 2, and 3) and then write all of the blocks (including the parity block) to the five disks above in parallel (highlighted in gray in the figure). Thus, full-stripe writes are the most efficient way for RAID-4 to write to disk.

Once we understand the full-stripe write, calculating the performance of sequential writes on RAID-4 is easy; the effective bandwidth is also $(N-1)\cdot S$ MB/s. Even though the parity disk is constantly in use during the operation, the client does not gain performance advantage from it.

Now let us analyze the performance of random reads. As you can also see from the figure above, a set of 1-block random reads will be spread across the data disks of the system but not the parity disk. Thus, the effective performance is: $(N-1)\cdot R$ MB/s.

Random writes, which we have saved for last, present the most interesting case for RAID-4. Imagine we wish to overwrite block 1 in the example above. We could just go ahead and overwrite it, but that would leave us with a problem: the parity block P0 would no longer accurately reflect the correct parity value of the stripe; in this example, P0 must also be updated. How can we update it both correctly and efficiently?

It turns out there are two methods. The first, known as **additive parity**, requires us to do the following. To compute the value of the new parity block, read in all of the other data blocks in the stripe in parallel (in the example, blocks 0, 2, and 3) and XOR those with the new block (1). The result is your new parity block. To complete the write, you can then write the new data and new parity to their respective disks, also in parallel.

The problem with this technique is that it scales with the number of disks, and thus in larger RAIDs requires a high number of reads to compute parity. Thus, the **subtractive parity** method.

For example, imagine this string of bits (4 data bits, one parity):

C0	C1	C2	C3	P
0	0	1	1	XOR(0,0,1,1) = 0

Let's imagine that we wish to overwrite bit C2 with a new value which we will call $C2_{new}$. The subtractive method works in three steps. First, we read in the old data at C2 ($C2_{old}$ = 1) and the old parity (P_{old} = 0).

Then, we compare the old data and the new data; if they are the same (e.g., $C2_{new}$ = $C2_{old}$), then we know the parity bit will also remain the same (i.e., P_{new} = P_{old}). If, however, they are different, then we must flip the old parity bit to the opposite of its current state, that is, if (P_{old} == 1), P_{new} will be set to 0; if (P_{old} == 0), P_{new} will be set to 1. We can express this whole mess neatly with XOR (where $\oplus$ is the XOR operator):

$$P_{new} = (C_{old} \oplus C_{new}) \oplus P_{old} \tag{38.1}$$

Because we are dealing with blocks, not bits, we perform this calculation over all the bits in the block (e.g., 4096 bytes in each block multiplied by 8 bits per byte). Thus, in most cases, the new block will be different than the old block and thus the new parity block will too.

You should now be able to figure out when we would use the additive parity calculation and when we would use the subtractive method. Think about how many disks would need to be in the system so that the additive method performs fewer I/Os than the subtractive method; what is the cross-over point?

For this performance analysis, let us assume we are using the subtractive method. Thus, for each write, the RAID has to perform 4 physical I/Os (two reads and two writes). Now imagine there are lots of writes submitted to the RAID; how many can RAID-4 perform in parallel? To understand, let us again look at the RAID-4 layout (Figure 38.6).

Disk 0	Disk 1	Disk 2	Disk 3	Disk 4
0	1	2	3	P0
*4	5	6	7	+P1
8	9	10	11	P2
12	*13	14	15	+P3

Figure 38.6: **Example: Writes To 4, 13, And Respective Parity Blocks**

Now imagine there were 2 small writes submitted to the RAID-4 at about the same time, to blocks 4 and 13 (marked with * in the diagram). The data for those disks is on disks 0 and 1, and thus the read and write to data could happen in parallel, which is good. The problem that arises is with the parity disk; both the requests have to read the related parity blocks for 4 and 13, parity blocks 1 and 3 (marked with +). Hopefully, the issue is now clear: the parity disk is a bottleneck under this type of workload; we sometimes thus call this the **small-write problem** for parity-based RAIDs. Thus, even though the data disks could be accessed in parallel, the parity disk prevents any parallelism from materializing; all writes to the system will be serialized because of the parity disk. Because the parity disk has to perform two I/Os (one read, one write) per logical I/O, we can compute the performance of small random writes in RAID-4 by computing the parity disk's performance on those two I/Os, and thus we achieve $(R/2)$ MB/s. RAID-4 throughput under random small writes is terrible; it does not improve as you add disks to the system.

We conclude by analyzing I/O latency in RAID-4. As you now know, a single read (assuming no failure) is just mapped to a single disk, and thus its latency is equivalent to the latency of a single disk request. The latency of a single write requires two reads and then two writes; the reads can happen in parallel, as can the writes, and thus total latency is about twice that of a single disk (with some differences because we have to wait for both reads to complete and thus get the worst-case positioning time, but then the updates don't incur seek cost and thus may be a better-than-average positioning cost).

38.7 RAID Level 5: Rotating Parity

To address the small-write problem (at least, partially), Patterson, Gibson, and Katz introduced RAID-5. RAID-5 works almost identically to RAID-4, except that it **rotates** the parity block across drives (Figure 38.7).

Disk 0	Disk 1	Disk 2	Disk 3	Disk 4
0	1	2	3	P0
5	6	7	P1	4
10	11	P2	8	9
15	P3	12	13	14
P4	16	17	18	19

Figure 38.7: **RAID-5 With Rotated Parity**

As you can see, the parity block for each stripe is now rotated across the disks, in order to remove the parity-disk bottleneck for RAID-4.

RAID-5 Analysis

Much of the analysis for RAID-5 is identical to RAID-4. For example, the effective capacity and failure tolerance of the two levels are identical. So are sequential read and write performance. The latency of a single request (whether a read or a write) is also the same as RAID-4.

Random read performance is a little better, because we can now utilize all disks. Finally, random write performance improves noticeably over RAID-4, as it allows for parallelism across requests. Imagine a write to block 1 and a write to block 10; this will turn into requests to disk 1 and disk 4 (for block 1 and its parity) and requests to disk 0 and disk 2 (for block 10 and its parity). Thus, they can proceed in parallel. In fact, we can generally assume that given a large number of random requests, we will be able to keep all the disks about evenly busy. If that is the case, then our total bandwidth for small writes will be $\frac{N}{4} \cdot R$ MB/s. The factor of four loss is due to the fact that each RAID-5 write still generates 4 total I/O operations, which is simply the cost of using parity-based RAID.

	RAID-0	RAID-1	RAID-4	RAID-5
Capacity	$N \cdot B$	$(N \cdot B)/2$	$(N-1) \cdot B$	$(N-1) \cdot B$
Reliability	0	1 (for sure) $\frac{N}{2}$ (if lucky)	1	1
Throughput				
Sequential Read	$N \cdot S$	$(N/2) \cdot S$	$(N-1) \cdot S$	$(N-1) \cdot S$
Sequential Write	$N \cdot S$	$(N/2) \cdot S$	$(N-1) \cdot S$	$(N-1) \cdot S$
Random Read	$N \cdot R$	$N \cdot R$	$(N-1) \cdot R$	$N \cdot R$
Random Write	$N \cdot R$	$(N/2) \cdot R$	$\frac{1}{2} \cdot R$	$\frac{N}{4} R$
Latency				
Read	T	T	T	T
Write	T	T	$2T$	$2T$

Figure 38.8: **RAID Capacity, Reliability, and Performance**

Because RAID-5 is basically identical to RAID-4 except in the few cases where it is better, it has almost completely replaced RAID-4 in the marketplace. The only place where it has not is in systems that know they will never perform anything other than a large write, thus avoiding the small-write problem altogether [HLM94]; in those cases, RAID-4 is sometimes used as it is slightly simpler to build.

38.8 RAID Comparison: A Summary

We now summarize our simplified comparison of RAID levels in Figure 38.8. Note that we have omitted a number of details to simplify our analysis. For example, when writing in a mirrored system, the average seek time is a little higher than when writing to just a single disk, because the seek time is the max of two seeks (one on each disk). Thus, random write performance to two disks will generally be a little less than random write performance of a single disk. Also, when updating the parity disk in RAID-4/5, the first read of the old parity will likely cause a full seek and rotation, but the second write of the parity will only result in rotation.

However, the comparison in Figure 38.8 does capture the essential differences, and is useful for understanding tradeoffs across RAID levels. For the latency analysis, we simply use T to represent the time that a request to a single disk would take.

To conclude, if you strictly want performance and do not care about reliability, striping is obviously best. If, however, you want random I/O performance and reliability, mirroring is the best; the cost you pay is in lost capacity. If capacity and reliability are your main goals, then RAID-5 is the winner; the cost you pay is in small-write performance. Finally, if you are always doing sequential I/O and want to maximize capacity, RAID-5 also makes the most sense.

38.9 Other Interesting RAID Issues

There are a number of other interesting ideas that one could (and perhaps should) discuss when thinking about RAID. Here are some things we might eventually write about.

For example, there are many other RAID designs, including Levels 2 and 3 from the original taxonomy, and Level 6 to tolerate multiple disk faults [C+04]. There is also what the RAID does when a disk fails; sometimes it has a **hot spare** sitting around to fill in for the failed disk. What happens to performance under failure, and performance during reconstruction of the failed disk? There are also more realistic fault models, to take into account **latent sector errors** or **block corruption** [B+08], and lots of techniques to handle such faults (see the data integrity chapter for details). Finally, you can even build RAID as a software layer: such **software RAID** systems are cheaper but have other problems, including the consistent-update problem [DAA05].

38.10 Summary

We have discussed RAID. RAID transforms a number of independent disks into a large, more capacious, and more reliable single entity; importantly, it does so transparently, and thus hardware and software above is relatively oblivious to the change.

There are many possible RAID levels to choose from, and the exact RAID level to use depends heavily on what is important to the end-user. For example, mirrored RAID is simple, reliable, and generally provides good performance but at a high capacity cost. RAID-5, in contrast, is reliable and better from a capacity standpoint, but performs quite poorly when there are small writes in the workload. Picking a RAID and setting its parameters (chunk size, number of disks, etc.) properly for a particular workload is challenging, and remains more of an art than a science.

References

[B+08] "An Analysis of Data Corruption in the Storage Stack"
Lakshmi N. Bairavasundaram, Garth R. Goodson, Bianca Schroeder, Andrea C. Arpaci-Dusseau, Remzi H. Arpaci-Dusseau
FAST '08, San Jose, CA, February 2008
Our own work analyzing how often disks actually corrupt your data. Not often, but sometimes! And thus something a reliable storage system must consider.

[BJ88] "Disk Shadowing"
D. Bitton and J. Gray
VLDB 1988
One of the first papers to discuss mirroring, herein called "shadowing".

[CL95] "Striping in a RAID level 5 disk array"
Peter M. Chen, Edward K. Lee
SIGMETRICS 1995
A nice analysis of some of the important parameters in a RAID-5 disk array.

[C+04] "Row-Diagonal Parity for Double Disk Failure Correction"
P. Corbett, B. English, A. Goel, T. Grcanac, S. Kleiman, J. Leong, S. Sankar
FAST '04, February 2004
Though not the first paper on a RAID system with two disks for parity, it is a recent and highly-understandable version of said idea. Read it to learn more.

[DAA05] "Journal-guided Resynchronization for Software RAID"
Timothy E. Denehy, A. Arpaci-Dusseau, R. Arpaci-Dusseau
FAST 2005
Our own work on the consistent-update problem. Here we solve it for Software RAID by integrating the journaling machinery of the file system above with the software RAID beneath it.

[HLM94] "File System Design for an NFS File Server Appliance"
Dave Hitz, James Lau, Michael Malcolm
USENIX Winter 1994, San Francisco, California, 1994
The sparse paper introducing a landmark product in storage, the write-anywhere file layout or WAFL file system that underlies the NetApp file server.

[K86] "Synchronized Disk Interleaving"
M.Y. Kim.
IEEE Transactions on Computers, Volume C-35: 11, November 1986
Some of the earliest work on RAID is found here.

[K88] "Small Disk Arrays - The Emerging Approach to High Performance"
F. Kurzweil.
Presentation at Spring COMPCON '88, March 1, 1988, San Francisco, California
Another early RAID reference.

[P+88] "Redundant Arrays of Inexpensive Disks"
D. Patterson, G. Gibson, R. Katz.
SIGMOD 1988
*This is considered **the** RAID paper, written by famous authors Patterson, Gibson, and Katz. The paper has since won many test-of-time awards and ushered in the RAID era, including the name RAID itself!*

[PB86] "Providing Fault Tolerance in Parallel Secondary Storage Systems"
A. Park and K. Balasubramaniam
Department of Computer Science, Princeton, CS-TR-O57-86, November 1986
Another early work on RAID.

[SG86] "Disk Striping"
K. Salem and H. Garcia-Molina.
IEEE International Conference on Data Engineering, 1986
And yes, another early RAID work. There are a lot of these, which kind of came out of the woodwork when the RAID paper was published in SIGMOD.

[S84] "Byzantine Generals in Action: Implementing Fail-Stop Processors"
F.B. Schneider.
ACM Transactions on Computer Systems, 2(2):145154, May 1984
Finally, a paper that is not about RAID! This paper is actually about how systems fail, and how to make something behave in a fail-stop manner.

Homework

This section introduces `raid.py`, a simple RAID simulator you can use to shore up your knowledge of how RAID systems work. See the README for details.

Questions

1. Use the simulator to perform some basic RAID mapping tests. Run with different levels (0, 1, 4, 5) and see if you can figure out the mappings of a set of requests. For RAID-5, see if you can figure out the difference between left-symmetric and left-asymmetric layouts. Use some different random seeds to generate different problems than above.
2. Do the same as the first problem, but this time vary the chunk size with `-C`. How does chunk size change the mappings?
3. Do the same as above, but use the `-r` flag to reverse the nature of each problem.
4. Now use the reverse flag but increase the size of each request with the `-S` flag. Try specifying sizes of 8k, 12k, and 16k, while varying the RAID level. What happens to the underlying I/O pattern when the size of the request increases? Make sure to try this with the sequential workload too (`-W sequential`); for what request sizes are RAID-4 and RAID-5 much more I/O efficient?
5. Use the timing mode of the simulator (`-t`) to estimate the performance of 100 random reads to the RAID, while varying the RAID levels, using 4 disks.
6. Do the same as above, but increase the number of disks. How does the performance of each RAID level scale as the number of disks increases?
7. Do the same as above, but use all writes (`-w 100`) instead of reads. How does the performance of each RAID level scale now? Can you do a rough estimate of the time it will take to complete the workload of 100 random writes?
8. Run the timing mode one last time, but this time with a sequential workload (`-W sequential`). How does the performance vary with RAID level, and when doing reads versus writes? How about when varying the size of each request? What size should you write to a RAID when using RAID-4 or RAID-5?

39

Interlude: Files and Directories

Thus far we have seen the development of two key operating system abstractions: the process, which is a virtualization of the CPU, and the address space, which is a virtualization of memory. In tandem, these two abstractions allow a program to run as if it is in its own private, isolated world; as if it has its own processor (or processors); as if it has its own memory. This illusion makes programming the system much easier and thus is prevalent today not only on desktops and servers but increasingly on all programmable platforms including mobile phones and the like.

In this section, we add one more critical piece to the virtualization puzzle: **persistent storage**. A persistent-storage device, such as a classic **hard disk drive** or a more modern **solid-state storage device**, stores information permanently (or at least, for a long time). Unlike memory, whose contents are lost when there is a power loss, a persistent-storage device keeps such data intact. Thus, the OS must take extra care with such a device: this is where users keep data that they really care about.

CRUX: HOW TO MANAGE A PERSISTENT DEVICE

How should the OS manage a persistent device? What are the APIs? What are the important aspects of the implementation?

Thus, in the next few chapters, we will explore critical techniques for managing persistent data, focusing on methods to improve performance and reliability. We begin, however, with an overview of the API: the interfaces you'll expect to see when interacting with a UNIX file system.

39.1 Files and Directories

Two key abstractions have developed over time in the virtualization of storage. The first is the **file**. A file is simply a linear array of bytes, each of which you can read or write. Each file has some kind of **low-level**

name, usually a number of some kind; often, the user is not aware of this name (as we will see). For historical reasons, the low-level name of a file is often referred to as its **inode number**. We'll be learning a lot more about inodes in future chapters; for now, just assume that each file has an inode number associated with it.

In most systems, the OS does not know much about the structure of the file (e.g., whether it is a picture, or a text file, or C code); rather, the responsibility of the file system is simply to store such data persistently on disk and make sure that when you request the data again, you get what you put there in the first place. Doing so is not as simple as it seems!

The second abstraction is that of a **directory**. A directory, like a file, also has a low-level name (i.e., an inode number), but its contents are quite specific: it contains a list of (user-readable name, low-level name) pairs. For example, let's say there is a file with the low-level name "10", and it is referred to by the user-readable name of "foo". The directory that "foo" resides in thus would have an entry ("foo", "10") that maps the user-readable name to the low-level name. Each entry in a directory refers to either files or other directories. By placing directories within other directories, users are able to build an arbitrary **directory tree** (or **directory hierarchy**), under which all files and directories are stored.

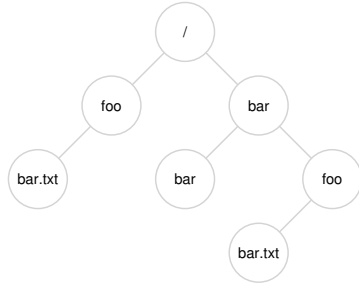


Figure 39.1: **An Example Directory Tree**

The directory hierarchy starts at a **root directory** (in UNIX-based systems, the root directory is simply referred to as `/`) and uses some kind of **separator** to name subsequent **sub-directories** until the desired file or directory is named. For example, if a user created a directory `foo` in the root directory `/`, and then created a file `bar.txt` in the directory `foo`, we could refer to the file by its **absolute pathname**, which in this case would be `/foo/bar.txt`. See Figure 39.1 for a more complex directory tree; valid directories in the example are `/`, `/foo`, `/bar`, `/bar/bar`, `/bar/foo` and valid files are `/foo/bar.txt` and `/bar/foo/bar.txt`. Directories and files can have the same name as long as they are in different locations in the file-system tree (e.g., there are two files named `bar.txt` in the figure, `/foo/bar.txt` and `/bar/foo/bar.txt`).

TIP: THINK CAREFULLY ABOUT NAMING
Naming is an important aspect of computer systems [SK09]. In UNIX systems, virtually everything that you can think of is named through the file system. Beyond just files, devices, pipes, and even processes [K84] can be found in what looks like a plain old file system. This uniformity of naming eases your conceptual model of the system, and makes the system simpler and more modular. Thus, whenever creating a system or interface, think carefully about what names you are using.

You may also notice that the file name in this example often has two parts: `bar` and `txt`, separated by a period. The first part is an arbitrary name, whereas the second part of the file name is usually used to indicate the **type** of the file, e.g., whether it is C code (e.g., `.c`), or an image (e.g., `.jpg`), or a music file (e.g., `.mp3`). However, this is usually just a **convention**: there is usually no enforcement that the data contained in a file named `main.c` is indeed C source code.

Thus, we can see one great thing provided by the file system: a convenient way to **name** all the files we are interested in. Names are important in systems as the first step to accessing any resource is being able to name it. In UNIX systems, the file system thus provides a unified way to access files on disk, USB stick, CD-ROM, many other devices, and in fact many other things, all located under the single directory tree.

39.2 The File System Interface

Let's now discuss the file system interface in more detail. We'll start with the basics of creating, accessing, and deleting files. You may think this is straightforward, but along the way we'll discover the mysterious call that is used to remove files, known as `unlink()`. Hopefully, by the end of this chapter, this mystery won't be so mysterious to you!

39.3 Creating Files

We'll start with the most basic of operations: creating a file. This can be accomplished with the `open` system call; by calling `open()` and passing it the `O_CREAT` flag, a program can create a new file. Here is some example code to create a file called "foo" in the current working directory.

```
int fd = open("foo", O_CREAT|O_WRONLY|O_TRUNC, S_IRUSR|S_IWUSR);
```

The routine `open()` takes a number of different flags. In this example, the second parameter creates the file (`O_CREAT`) if it does not exist, ensures that the file can only be written to (`O_WRONLY`), and, if the file already exists, truncates it to a size of zero bytes thus removing any existing content (`O_TRUNC`). The third parameter specifies permissions, in this case making the file readable and writable by the owner.

ASIDE: THE CREAT() SYSTEM CALL

The older way of creating a file is to call `creat()`, as follows:

```
int fd = creat("foo"); // option: add second flag to set permissions
```

You can think of `creat()` as `open()` with the following flags: `O_CREAT | O_WRONLY | O_TRUNC`. Because `open()` can create a file, the usage of `creat()` has somewhat fallen out of favor (indeed, it could just be implemented as a library call to `open()`); however, it does hold a special place in UNIX lore. Specifically, when Ken Thompson was asked what he would do differently if he were redesigning UNIX, he replied: "I'd spell creat with an e."

One important aspect of `open()` is what it returns: a **file descriptor**. A file descriptor is just an integer, private per process, and is used in UNIX systems to access files; thus, once a file is opened, you use the file descriptor to read or write the file, assuming you have permission to do so. In this way, a file descriptor is a **capability** [L84], i.e., an opaque handle that gives you the power to perform certain operations. Another way to think of a file descriptor is as a pointer to an object of type file; once you have such an object, you can call other "methods" to access the file, like `read()` and `write()`. We'll see just how a file descriptor is used below.

39.4 Reading and Writing Files

Once we have some files, of course we might like to read or write them. Let's start by reading an existing file. If we were typing at a command line, we might just use the program `cat` to dump the contents of the file to the screen.

```
prompt> echo hello > foo
prompt> cat foo
hello
prompt>
```

In this code snippet, we redirect the output of the program `echo` to the file `foo`, which then contains the word "hello" in it. We then use `cat` to see the contents of the file. But how does the `cat` program access the file `foo`?

To find this out, we'll use an incredibly useful tool to trace the system calls made by a program. On Linux, the tool is called **strace**; other systems have similar tools (see **dtruss** on a Mac, or **truss** on some older UNIX variants). What `strace` does is trace every system call made by a program while it runs, and dump the trace to the screen for you to see.

TIP: USE STRACE (AND SIMILAR TOOLS)

The `strace` tool provides an awesome way to see what programs are up to. By running it, you can trace which system calls a program makes, see the arguments and return codes, and generally get a very good idea of what is going on.

The tool also takes some arguments which can be quite useful. For example, `-f` follows any fork'd children too; `-t` reports the time of day at each call; `-e trace=open,close,read,write` only traces calls to those system calls and ignores all others. There are many more powerful flags — read the man pages and find out how to harness this wonderful tool.

Here is an example of using `strace` to figure out what `cat` is doing (some calls removed for readability):

```
prompt> strace cat foo
...
open("foo", O_RDONLY|O_LARGEFILE)       = 3
read(3, "hello\n", 4096)                = 6
write(1, "hello\n", 6)                  = 6
hello
read(3, "", 4096)                       = 0
close(3)                                = 0
...
prompt>
```

The first thing that `cat` does is open the file for reading. A couple of things we should note about this; first, that the file is only opened for reading (not writing), as indicated by the `O_RDONLY` flag; second, that the 64-bit offset be used (`O_LARGEFILE`); third, that the call to `open()` succeeds and returns a file descriptor, which has the value of 3.

Why does the first call to `open()` return 3, not 0 or perhaps 1 as you might expect? As it turns out, each running process already has three files open, standard input (which the process can read to receive input), standard output (which the process can write to in order to dump information to the screen), and standard error (which the process can write error messages to). These are represented by file descriptors 0, 1, and 2, respectively. Thus, when you first open another file (as `cat` does above), it will almost certainly be file descriptor 3.

After the open succeeds, `cat` uses the `read()` system call to repeatedly read some bytes from a file. The first argument to `read()` is the file descriptor, thus telling the file system which file to read; a process can of course have multiple files open at once, and thus the descriptor enables the operating system to know which file a particular read refers to. The second argument points to a buffer where the result of the `read()` will be placed; in the system-call trace above, strace shows the results of the read in this spot ("hello"). The third argument is the size of the buffer, which

in this case is 4 KB. The call to `read()` returns successfully as well, here returning the number of bytes it read (6, which includes 5 for the letters in the word "hello" and one for an end-of-line marker).

At this point, you see another interesting result of the strace: a single call to the `write()` system call, to the file descriptor 1. As we mentioned above, this descriptor is known as the standard output, and thus is used to write the word "hello" to the screen as the program `cat` is meant to do. But does it call `write()` directly? Maybe (if it is highly optimized). But if not, what `cat` might do is call the library routine `printf()`; internally, `printf()` figures out all the formatting details passed to it, and eventually calls write on the standard output to print the results to the screen.

The `cat` program then tries to read more from the file, but since there are no bytes left in the file, the `read()` returns 0 and the program knows that this means it has read the entire file. Thus, the program calls `close()` to indicate that it is done with the file "foo", passing in the corresponding file descriptor. The file is thus closed, and the reading of it thus complete.

Writing a file is accomplished via a similar set of steps. First, a file is opened for writing, then the `write()` system call is called, perhaps repeatedly for larger files, and then `close()`. Use `strace` to trace writes to a file, perhaps of a program you wrote yourself, or by tracing the `dd` utility, e.g., `dd if=foo of=bar`.

39.5 Reading And Writing, But Not Sequentially

Thus far, we've discussed how to read and write files, but all access has been **sequential**; that is, we have either read a file from the beginning to the end, or written a file out from beginning to end.

Sometimes, however, it is useful to be able to read or write to a specific offset within a file; for example, if you build an index over a text document, and use it to look up a specific word, you may end up reading from some **random** offsets within the document. To do so, we will use the `lseek()` system call. Here is the function prototype:

```
off_t lseek(int fildes, off_t offset, int whence);
```

The first argument is familiar (a file descriptor). The second argument is the `offset`, which positions the **file offset** to a particular location within the file. The third argument, called `whence` for historical reasons, determines exactly how the seek is performed. From the man page:

```
If whence is SEEK_SET, the offset is set to offset bytes.
If whence is SEEK_CUR, the offset is set to its current
location plus offset bytes.
If whence is SEEK_END, the offset is set to the size of
the file plus offset bytes.
```

As you can tell from this description, for each file a process opens, the OS tracks a "current" offset, which determines where the next read or

ASIDE: **CALLING LSEEK() DOES NOT PERFORM A DISK SEEK**
The poorly-named system call `lseek()` confuses many a student trying to understand disks and how the file systems atop them work. Do not confuse the two! The `lseek()` call simply changes a variable in OS memory that tracks, for a particular process, at which offset to which its next read or write will start. A disk seek occurs when a read or write issued to the disk is not on the same track as the last read or write, and thus necessitates a head movement. Making this even more confusing is the fact that calling `lseek()` to read or write from/to random parts of a file, and then reading/writing to those random parts, will indeed lead to more disk seeks. Thus, calling `lseek()` can certainly lead to a seek in an upcoming read or write, but absolutely does not cause any disk I/O to occur itself.

write will begin reading from or writing to within the file. Thus, part of the abstraction of an open file is that it has a current offset, which is updated in one of two ways. The first is when a read or write of N bytes takes place, N is added to the current offset; thus each read or write *implicitly* updates the offset. The second is *explicitly* with `lseek`, which changes the offset as specified above.

Note that this call `lseek()` has nothing to do with the **seek** operation of a disk, which moves the disk arm. The call to `lseek()` simply changes the value of a variable within the kernel; when the I/O is performed, depending on where the disk head is, the disk may or may not perform an actual seek to fulfill the request.

39.6 Writing Immediately with `fsync()`

Most times when a program calls `write()`, it is just telling the file system: please write this data to persistent storage, at some point in the future. The file system, for performance reasons, will **buffer** such writes in memory for some time (say 5 seconds, or 30); at that later point in time, the write(s) will actually be issued to the storage device. From the perspective of the calling application, writes seem to complete quickly, and only in rare cases (e.g., the machine crashes after the `write()` call but before the write to disk) will data be lost.

However, some applications require something more than this eventual guarantee. For example, in a database management system (DBMS), development of a correct recovery protocol requires the ability to force writes to disk from time to time.

To support these types of applications, most file systems provide some additional control APIs. In the UNIX world, the interface provided to applications is known as `fsync(int fd)`. When a process calls `fsync()` for a particular file descriptor, the file system responds by forcing all **dirty** (i.e., not yet written) data to disk, for the file referred to by the specified

file descriptor. The `fsync()` routine returns once all of these writes are complete.

Here is a simple example of how to use `fsync()`. The code opens the file `foo`, writes a single chunk of data to it, and then calls `fsync()` to ensure the writes are forced immediately to disk. Once the `fsync()` returns, the application can safely move on, knowing that the data has been persisted (if `fsync()` is correctly implemented, that is).

```
int fd = open("foo", O_CREAT|O_WRONLY|O_TRUNC, S_IRUSR|S_IWUSR);
assert(fd > -1);
int rc = write(fd, buffer, size);
assert(rc == size);
rc = fsync(fd);
assert(rc == 0);
```

Interestingly, this sequence does not guarantee everything that you might expect; in some cases, you also need to `fsync()` the directory that contains the file `foo`. Adding this step ensures not only that the file itself is on disk, but that the file, if newly created, also is durably a part of the directory. Not surprisingly, this type of detail is often overlooked, leading to many application-level bugs [P+13,P+14].

39.7 Renaming Files

Once we have a file, it is sometimes useful to be able to give a file a different name. When typing at the command line, this is accomplished with `mv` command; in this example, the file `foo` is renamed `bar`:

```
prompt> mv foo bar
```

Using `strace`, we can see that `mv` uses the system call `rename(char *old, char *new)`, which takes precisely two arguments: the original name of the file (`old`) and the new name (`new`).

One interesting guarantee provided by the `rename()` call is that it is (usually) implemented as an **atomic** call with respect to system crashes; if the system crashes during the renaming, the file will either be named the old name or the new name, and no odd in-between state can arise. Thus, `rename()` is critical for supporting certain kinds of applications that require an atomic update to file state.

Let's be a little more specific here. Imagine that you are using a file editor (e.g., emacs), and you insert a line into the middle of a file. The file's name, for the example, is `foo.txt`. The way the editor might update the file to guarantee that the new file has the original contents plus the line inserted is as follows (ignoring error-checking for simplicity):

```
int fd = open("foo.txt.tmp", O_WRONLY|O_CREAT|O_TRUNC,
              S_IRUSR|S_IWUSR);
write(fd, buffer, size); // write out new version of file
fsync(fd);
close(fd);
rename("foo.txt.tmp", "foo.txt");
```

What the editor does in this example is simple: write out the new version of the file under a temporary name (`foo.txt.tmp`), force it to disk with `fsync()`, and then, when the application is certain the new file metadata and contents are on the disk, rename the temporary file to the original file's name. This last step atomically swaps the new file into place, while concurrently deleting the old version of the file, and thus an atomic file update is achieved.

39.8 Getting Information About Files

Beyond file access, we expect the file system to keep a fair amount of information about each file it is storing. We generally call such data about files **metadata**. To see the metadata for a certain file, we can use the `stat()` or `fstat()` system calls. These calls take a pathname (or file descriptor) to a file and fill in a `stat` structure as seen here:

```
struct stat {
    dev_t     st_dev;     /* ID of device containing file */
    ino_t     st_ino;     /* inode number */
    mode_t    st_mode;    /* protection */
    nlink_t   st_nlink;   /* number of hard links */
    uid_t     st_uid;     /* user ID of owner */
    gid_t     st_gid;     /* group ID of owner */
    dev_t     st_rdev;    /* device ID (if special file) */
    off_t     st_size;    /* total size, in bytes */
    blksize_t st_blksize; /* blocksize for filesystem I/O */
    blkcnt_t  st_blocks;  /* number of blocks allocated */
    time_t    st_atime;   /* time of last access */
    time_t    st_mtime;   /* time of last modification */
    time_t    st_ctime;   /* time of last status change */
};
```

You can see that there is a lot of information kept about each file, including its size (in bytes), its low-level name (i.e., inode number), some ownership information, and some information about when the file was accessed or modified, among other things. To see this information, you can use the command line tool `stat`:

```
prompt> echo hello > file
prompt> stat file
  File: 'file'
  Size: 6 Blocks: 8          IO Block: 4096   regular file
Device: 811h/2065d Inode: 67158084    Links: 1
Access: (0640/-rw-r-----) Uid: (30686/ remzi) Gid: (30686/ remzi)
Access: 2011-05-03 15:50:20.157594748 -0500
Modify: 2011-05-03 15:50:20.157594748 -0500
Change: 2011-05-03 15:50:20.157594748 -0500
```

As it turns out, each file system usually keeps this type of information in a structure called an **inode**[1]. We'll be learning a lot more about inodes when we talk about file system implementation. For now, you should just think of an inode as a persistent data structure kept by the file system that has information like we see above inside of it.

39.9 Removing Files

At this point, we know how to create files and access them, either sequentially or not. But how do you delete files? If you've used UNIX, you probably think you know: just run the program `rm`. But what system call does `rm` use to remove a file?

Let's use our old friend `strace` again to find out. Here we remove that pesky file "foo":

```
prompt> strace rm foo
...
unlink("foo")                           = 0
...
```

We've removed a bunch of unrelated cruft from the traced output, leaving just a single call to the mysteriously-named system call `unlink()`. As you can see, `unlink()` just takes the name of the file to be removed, and returns zero upon success. But this leads us to a great puzzle: why is this system call named "unlink"? Why not just "remove" or "delete". To understand the answer to this puzzle, we must first understand more than just files, but also directories.

39.10 Making Directories

Beyond files, a set of directory-related system calls enable you to make, read, and delete directories. Note you can never write to a directory directly; because the format of the directory is considered file system metadata, you can only update a directory indirectly by, for example, creating files, directories, or other object types within it. In this way, the file system makes sure that the contents of the directory always are as expected.

To create a directory, a single system call, `mkdir()`, is available. The eponymous `mkdir` program can be used to create such a directory. Let's take a look at what happens when we run the `mkdir` program to make a simple directory called `foo`:

```
prompt> strace mkdir foo
...
mkdir("foo", 0777)                      = 0
...
prompt>
```

[1] Some file systems call these structures similar, but slightly different, names, such as dnodes; the basic idea is similar however.

TIP: BE WARY OF POWERFUL COMMANDS

The program `rm` provides us with a great example of powerful commands, and how sometimes too much power can be a bad thing. For example, to remove a bunch of files at once, you can type something like:

```
prompt> rm *
```

where the `*` will match all files in the current directory. But sometimes you want to also delete the directories too, and in fact all of their contents. You can do this by telling `rm` to recursively descend into each directory, and remove its contents too:

```
prompt> rm -rf *
```

Where you get into trouble with this small string of characters is when you issue the command, accidentally, from the root directory of a file system, thus removing every file and directory from it. Oops!

Thus, remember the double-edged sword of powerful commands; while they give you the ability to do a lot of work with a small number of keystrokes, they also can quickly and readily do a great deal of harm.

When such a directory is created, it is considered "empty", although it does have a bare minimum of contents. Specifically, an empty directory has two entries: one entry that refers to itself, and one entry that refers to its parent. The former is referred to as the "." (dot) directory, and the latter as ".." (dot-dot). You can see these directories by passing a flag (`-a`) to the program `ls`:

```
prompt> ls -a
./  ../
prompt> ls -al
total 8
drwxr-x---  2 remzi remzi    6 Apr 30 16:17 ./
drwxr-x--- 26 remzi remzi 4096 Apr 30 16:17 ../
```

39.11 Reading Directories

Now that we've created a directory, we might wish to read one too. Indeed, that is exactly what the program `ls` does. Let's write our own little tool like `ls` and see how it is done.

Instead of just opening a directory as if it were a file, we instead use a new set of calls. Below is an example program that prints the contents of a directory. The program uses three calls, `opendir()`, `readdir()`, and `closedir()`, to get the job done, and you can see how simple the interface is; we just use a simple loop to read one directory entry at a time, and print out the name and inode number of each file in the directory.

```
int main(int argc, char *argv[]) {
    DIR *dp = opendir(".");
    assert(dp != NULL);
    struct dirent *d;
    while ((d = readdir(dp)) != NULL) {
        printf("%lu %s\n", (unsigned long) d->d_ino, d->d_name);
    }
    closedir(dp);
    return 0;
}
```

The declaration below shows the information available within each directory entry in the `struct dirent` data structure:

```
struct dirent {
    char           d_name[256]; /* filename */
    ino_t          d_ino;       /* inode number */
    off_t          d_off;       /* offset to the next dirent */
    unsigned short d_reclen;    /* length of this record */
    unsigned char  d_type;      /* type of file */
};
```

Because directories are light on information (basically, just mapping the name to the inode number, along with a few other details), a program may want to call `stat()` on each file to get more information on each, such as its length or other detailed information. Indeed, this is exactly what `ls` does when you pass it the `-l` flag; try `strace` on `ls` with and without that flag to see for yourself.

39.12 Deleting Directories

Finally, you can delete a directory with a call to `rmdir()` (which is used by the program of the same name, `rmdir`). Unlike file deletion, however, removing directories is more dangerous, as you could potentially delete a large amount of data with a single command. Thus, `rmdir()` has the requirement that the directory be empty (i.e., only has "." and ".." entries) before it is deleted. If you try to delete a non-empty directory, the call to `rmdir()` simply will fail.

39.13 Hard Links

We now come back to the mystery of why removing a file is performed via `unlink()`, by understanding a new way to make an entry in the file system tree, through a system call known as `link()`. The `link()` system call takes two arguments, an old pathname and a new one; when you "link" a new file name to an old one, you essentially create another way to refer to the same file. The command-line program `ln` is used to do this, as we see in this example:

```
prompt> echo hello > file
prompt> cat file
hello
prompt> ln file file2
prompt> cat file2
hello
```

Here we created a file with the word "hello" in it, and called the file `file`[2]. We then create a hard link to that file using the `ln` program. After this, we can examine the file by either opening `file` or `file2`.

The way `link` works is that it simply creates another name in the directory you are creating the link to, and refers it to the *same* inode number (i.e., low-level name) of the original file. The file is not copied in any way; rather, you now just have two human names (`file` and `file2`) that both refer to the same file. We can even see this in the directory itself, by printing out the inode number of each file:

```
prompt> ls -i file file2
67158084 file
67158084 file2
prompt>
```

By passing the `-i` flag to `ls`, it prints out the inode number of each file (as well as the file name). And thus you can see what link really has done: just make a new reference to the same exact inode number (67158084 in this example).

By now you might be starting to see why `unlink()` is called `unlink()`. When you create a file, you are really doing *two* things. First, you are making a structure (the inode) that will track virtually all relevant information about the file, including its size, where its blocks are on disk, and so forth. Second, you are *linking* a human-readable name to that file, and putting that link into a directory.

After creating a hard link to a file, to the file system, there is no difference between the original file name (`file`) and the newly created file name (`file2`); indeed, they are both just links to the underlying metadata about the file, which is found in inode number 67158084.

Thus, to remove a file from the file system, we call `unlink()`. In the example above, we could for example remove the file named `file`, and still access the file without difficulty:

```
prompt> rm file
removed 'file'
prompt> cat file2
hello
```

The reason this works is because when the file system unlinks file, it checks a **reference count** within the inode number. This reference count

[2]Note how creative the authors of this book are. We also used to have a cat named "Cat" (true story). However, she died, and we now have a hamster named "Hammy." Update: Hammy is now dead too. The pet bodies are piling up.

(sometimes called the **link count**) allows the file system to track how many different file names have been linked to this particular inode. When `unlink()` is called, it removes the "link" between the human-readable name (the file that is being deleted) to the given inode number, and decrements the reference count; only when the reference count reaches zero does the file system also free the inode and related data blocks, and thus truly "delete" the file.

You can see the reference count of a file using `stat()` of course. Let's see what it is when we create and delete hard links to a file. In this example, we'll create three links to the same file, and then delete them. Watch the link count!

```
prompt> echo hello > file
prompt> stat file
... Inode: 67158084    Links: 1 ...
prompt> ln file file2
prompt> stat file
... Inode: 67158084    Links: 2 ...
prompt> stat file2
... Inode: 67158084    Links: 2 ...
prompt> ln file2 file3
prompt> stat file
... Inode: 67158084    Links: 3 ...
prompt> rm file
prompt> stat file2
... Inode: 67158084    Links: 2 ...
prompt> rm file2
prompt> stat file3
... Inode: 67158084    Links: 1 ...
prompt> rm file3
```

39.14 Symbolic Links

There is one other type of link that is really useful, and it is called a **symbolic link** or sometimes a **soft link**. As it turns out, hard links are somewhat limited: you can't create one to a directory (for fear that you will create a cycle in the directory tree); you can't hard link to files in other disk partitions (because inode numbers are only unique within a particular file system, not across file systems); etc. Thus, a new type of link called the symbolic link was created.

To create such a link, you can use the same program `ln`, but with the `-s` flag. Here is an example:

```
prompt> echo hello > file
prompt> ln -s file file2
prompt> cat file2
hello
```

As you can see, creating a soft link looks much the same, and the original file can now be accessed through the file name `file` as well as the symbolic link name `file2`.

However, beyond this surface similarity, symbolic links are actually quite different from hard links. The first difference is that a symbolic link is actually a file itself, of a different type. We've already talked about regular files and directories; symbolic links are a third type the file system knows about. A `stat` on the symlink reveals all:

```
prompt> stat file
 ... regular file ...
prompt> stat file2
 ... symbolic link ...
```

Running `ls` also reveals this fact. If you look closely at the first character of the long-form of the output from `ls`, you can see that the first character in the left-most column is a `-` for regular files, a `d` for directories, and an `l` for soft links. You can also see the size of the symbolic link (4 bytes in this case), as well as what the link points to (the file named `file`).

```
prompt> ls -al
drwxr-x---  2 remzi remzi   29 May  3 19:10 ./
drwxr-x--- 27 remzi remzi 4096 May  3 15:14 ../
-rw-r-----  1 remzi remzi    6 May  3 19:10 file
lrwxrwxrwx  1 remzi remzi    4 May  3 19:10 file2 -> file
```

The reason that `file2` is 4 bytes is because the way a symbolic link is formed is by holding the pathname of the linked-to file as the data of the link file. Because we've linked to a file named `file`, our link file `file2` is small (4 bytes). If we link to a longer pathname, our link file would be bigger:

```
prompt> echo hello > alongerfilename
prompt> ln -s alongerfilename file3
prompt> ls -al alongerfilename file3
-rw-r----- 1 remzi remzi  6 May  3 19:17 alongerfilename
lrwxrwxrwx 1 remzi remzi 15 May  3 19:17 file3 -> alongerfilename
```

Finally, because of the way symbolic links are created, they leave the possibility for what is known as a **dangling reference**:

```
prompt> echo hello > file
prompt> ln -s file file2
prompt> cat file2
hello
prompt> rm file
prompt> cat file2
cat: file2: No such file or directory
```

As you can see in this example, quite unlike hard links, removing the original file named `file` causes the link to point to a pathname that no longer exists.

39.15 Making and Mounting a File System

We've now toured the basic interfaces to access files, directories, and certain types of special types of links. But there is one more topic we should discuss: how to assemble a full directory tree from many underlying file systems. This task is accomplished via first making file systems, and then mounting them to make their contents accessible.

To make a file system, most file systems provide a tool, usually referred to as `mkfs` (pronounced "make fs"), that performs exactly this task. The idea is as follows: give the tool, as input, a device (such as a disk partition, e.g., `/dev/sda1`) a file system type (e.g., ext3), and it simply writes an empty file system, starting with a root directory, onto that disk partition. And mkfs said, let there be a file system!

However, once such a file system is created, it needs to be made accessible within the uniform file-system tree. This task is achieved via the `mount` program (which makes the underlying system call `mount()` to do the real work). What mount does, quite simply is take an existing directory as a target **mount point** and essentially paste a new file system onto the directory tree at that point.

An example here might be useful. Imagine we have an unmounted ext3 file system, stored in device partition `/dev/sda1`, that has the following contents: a root directory which contains two sub-directories, `a` and `b`, each of which in turn holds a single file named `foo`. Let's say we wish to mount this file system at the mount point `/home/users`. We would type something like this:

```
prompt> mount -t ext3 /dev/sda1 /home/users
```

If successful, the mount would thus make this new file system available. However, note how the new file system is now accessed. To look at the contents of the root directory, we would use `ls` like this:

```
prompt> ls /home/users/
a b
```

As you can see, the pathname `/home/users/` now refers to the root of the newly-mounted directory. Similarly, we could access directories `a` and `b` with the pathnames `/home/users/a` and `/home/users/b`. Finally, the files named `foo` could be accessed via `/home/users/a/foo` and `/home/users/b/foo`. And thus the beauty of mount: instead of having a number of separate file systems, mount unifies all file systems into one tree, making naming uniform and convenient.

To see what is mounted on your system, and at which points, simply run the `mount` program. You'll see something like this:

```
/dev/sda1 on / type ext3 (rw)
proc on /proc type proc (rw)
sysfs on /sys type sysfs (rw)
/dev/sda5 on /tmp type ext3 (rw)
/dev/sda7 on /var/vice/cache type ext3 (rw)
tmpfs on /dev/shm type tmpfs (rw)
AFS on /afs type afs (rw)
```

This crazy mix shows that a whole number of different file systems, including ext3 (a standard disk-based file system), the proc file system (a file system for accessing information about current processes), tmpfs (a file system just for temporary files), and AFS (a distributed file system) are all glued together onto this one machine's file-system tree.

39.16 Summary

The file system interface in UNIX systems (and indeed, in any system) is seemingly quite rudimentary, but there is a lot to understand if you wish to master it. Nothing is better, of course, than simply using it (a lot). So please do so! Of course, read more; as always, Stevens [SR05] is the place to begin.

We've toured the basic interfaces, and hopefully understood a little bit about how they work. Even more interesting is how to implement a file system that meets the needs of the API, a topic we will delve into in great detail next.

References

[K84] "Processes as Files"
Tom J. Killian
USENIX, June 1984
The paper that introduced the /proc file system, where each process can be treated as a file within a pseudo file system. A clever idea that you can still see in modern UNIX systems.

[L84] "Capability-Based Computer Systems"
Henry M. Levy
Digital Press, 1984
Available: http://homes.cs.washington.edu/~levy/capabook
An excellent overview of early capability-based systems.

[P+13] "Towards Efficient, Portable Application-Level Consistency"
Thanumalayan S. Pillai, Vijay Chidambaram, Joo-Young Hwang, Andrea C. Arpaci-Dusseau, and Remzi H. Arpaci-Dusseau
HotDep '13, November 2013
Our own work that shows how readily applications can make mistakes in committing data to disk; in particular, assumptions about the file system creep into applications and thus make the applications work correctly only if they are running on a specific file system.

[P+14] "All File Systems Are Not Created Equal:
On the Complexity of Crafting Crash-Consistent Applications" Thanumalayan S. Pillai, Vijay Chidambaram, Ramnatthan Alagappan, Samer Al-Kiswany, Andrea C. Arpaci-Dusseau, and Remzi H. Arpaci-Dusseau
OSDI '14, Broomfield, Colorado
The full conference paper on this topic – with many more details and interesting tidbits than the first workshop paper above.

[SK09] "Principles of Computer System Design"
Jerome H. Saltzer and M. Frans Kaashoek
Morgan-Kaufmann, 2009
This tour de force of systems is a must-read for anybody interested in the field. It's how they teach systems at MIT. Read it once, and then read it a few more times to let it all soak in.

[SR05] "Advanced Programming in the UNIX Environment"
W. Richard Stevens and Stephen A. Rago
Addison-Wesley, 2005
We have probably referenced this book a few hundred thousand times. It is that useful to you, if you care to become an awesome systems programmer.

Homework

In this homework, we'll just familiarize ourselves with how the APIs described in the chapter work. To do so, you'll just write a few different programs, mostly based on various UNIX utilities.

Questions

1. **Stat:** Write your own version of the command line program `stat`, which simply calls the `stat()` system call on a given file or directory. Print out file size, number of blocks allocated, reference (link) count, and so forth. What is the link count of a directory, as the number of entries in the directory changes? Useful interfaces: `stat()`
2. **List Files:** Write a program that lists files in the given directory. When called without any arguments, the program should just print the file names. When invoked with the `-l` flag, the program should print out information about each file, such as the owner, group, permissions, and other information obtained from the `stat()` system call. The program should take one additional argument, which is the directory to read, e.g., `myls -l directory`. If no directory is given, the program should just use the current working directory. Useful interfaces: `stat()`, `opendir()`, `readdir()`, `getcwd()`.
3. **Tail:** Write a program that prints out the last few lines of a file. The program should be efficient, in that it seeks to near the end of the file, reads in a block of data, and then goes backwards until it finds the requested number of lines; at this point, it should print out those lines from beginning to the end of the file. To invoke the program, one should type: `mytail -n file`, where `n` is the number of lines at the end of the file to print. Useful interfaces: `stat()`, `lseek()`, `open()`, `read()`, `close()`.
4. **Recursive Search:** Write a program that prints out the names of each file and directory in the file system tree, starting at a given point in the tree. For example, when run without arguments, the program should start with the current working directory and print its contents, as well as the contents of any sub-directories, etc., until the entire tree, root at the CWD, is printed. If given a single argument (of a directory name), use that as the root of the tree instead. Refine your recursive search with more fun options, similar to the powerful `find` command line tool. Useful interfaces: you figure it out.

40

File System Implementation

In this chapter, we introduce a simple file system implementation, known as **vsfs** (the **Very Simple File System**). This file system is a simplified version of a typical UNIX file system and thus serves to introduce some of the basic on-disk structures, access methods, and various policies that you will find in many file systems today.

The file system is pure software; unlike our development of CPU and memory virtualization, we will not be adding hardware features to make some aspect of the file system work better (though we will want to pay attention to device characteristics to make sure the file system works well). Because of the great flexibility we have in building a file system, many different ones have been built, literally from AFS (the Andrew File System) [H+88] to ZFS (Sun's Zettabyte File System) [B07]. All of these file systems have different data structures and do some things better or worse than their peers. Thus, the way we will be learning about file systems is through case studies: first, a simple file system (vsfs) in this chapter to introduce most concepts, and then a series of studies of real file systems to understand how they can differ in practice.

THE CRUX: HOW TO IMPLEMENT A SIMPLE FILE SYSTEM
How can we build a simple file system? What structures are needed on the disk? What do they need to track? How are they accessed?

40.1 The Way To Think

To think about file systems, we usually suggest thinking about two different aspects of them; if you understand both of these aspects, you probably understand how the file system basically works.

The first is the **data structures** of the file system. In other words, what types of on-disk structures are utilized by the file system to organize its data and metadata? The first file systems we'll see (including vsfs below) employ simple structures, like arrays of blocks or other objects, whereas

ASIDE: **MENTAL MODELS OF FILE SYSTEMS**
As we've discussed before, mental models are what you are really trying to develop when learning about systems. For file systems, your mental model should eventually include answers to questions like: what on-disk structures store the file system's data and metadata? What happens when a process opens a file? Which on-disk structures are accessed during a read or write? By working on and improving your mental model, you develop an abstract understanding of what is going on, instead of just trying to understand the specifics of some file-system code (though that is also useful, of course!).

more sophisticated file systems, like SGI's XFS, use more complicated tree-based structures [S+96].

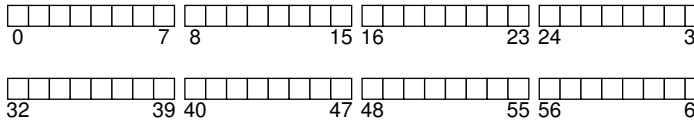
The second aspect of a file system is its **access methods**. How does it map the calls made by a process, such as `open()`, `read()`, `write()`, etc., onto its structures? Which structures are read during the execution of a particular system call? Which are written? How efficiently are all of these steps performed?

If you understand the data structures and access methods of a file system, you have developed a good mental model of how it truly works, a key part of the systems mindset. Try to work on developing your mental model as we delve into our first implementation.

40.2 Overall Organization

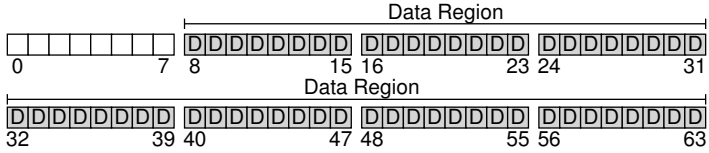
We now develop the overall on-disk organization of the data structures of the vsfs file system. The first thing we'll need to do is divide the disk into **blocks**; simple file systems use just one block size, and that's exactly what we'll do here. Let's choose a commonly-used size of 4 KB.

Thus, our view of the disk partition where we're building our file system is simple: a series of blocks, each of size 4 KB. The blocks are addressed from 0 to $N - 1$, in a partition of size N 4-KB blocks. Assume we have a really small disk, with just 64 blocks:



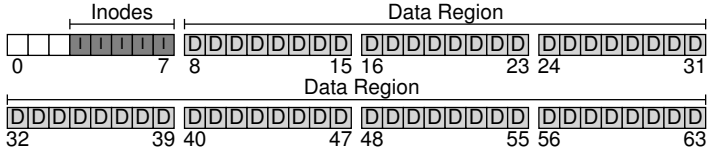
Let's now think about what we need to store in these blocks to build a file system. Of course, the first thing that comes to mind is user data. In fact, most of the space in any file system is (and should be) user data. Let's call the region of the disk we use for user data the **data region**, and,

again for simplicity, reserve a fixed portion of the disk for these blocks, say the last 56 of 64 blocks on the disk:



As we learned about (a little) last chapter, the file system has to track information about each file. This information is a key piece of **metadata**, and tracks things like which data blocks (in the data region) comprise a file, the size of the file, its owner and access rights, access and modify times, and other similar kinds of information. To store this information, file systems usually have a structure called an **inode** (we'll read more about inodes below).

To accommodate inodes, we'll need to reserve some space on the disk for them as well. Let's call this portion of the disk the **inode table**, which simply holds an array of on-disk inodes. Thus, our on-disk image now looks like this picture, assuming that we use 5 of our 64 blocks for inodes (denoted by I's in the diagram):

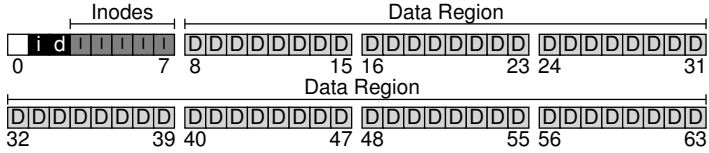


We should note here that inodes are typically not that big, for example 128 or 256 bytes. Assuming 256 bytes per inode, a 4-KB block can hold 16 inodes, and our file system above contains 80 total inodes. In our simple file system, built on a tiny 64-block partition, this number represents the maximum number of files we can have in our file system; however, do note that the same file system, built on a larger disk, could simply allocate a larger inode table and thus accommodate more files.

Our file system thus far has data blocks (D), and inodes (I), but a few things are still missing. One primary component that is still needed, as you might have guessed, is some way to track whether inodes or data blocks are free or allocated. Such **allocation structures** are thus a requisite element in any file system.

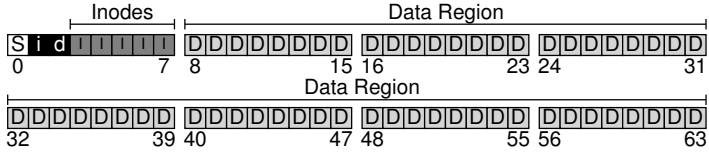
Many allocation-tracking methods are possible, of course. For example, we could use a **free list** that points to the first free block, which then points to the next free block, and so forth. We instead choose a simple and popular structure known as a **bitmap**, one for the data region (the **data bitmap**), and one for the inode table (the **inode bitmap**). A bitmap is a

simple structure: each bit is used to indicate whether the corresponding object/block is free (0) or in-use (1). And thus our new on-disk layout, with an inode bitmap (i) and a data bitmap (d):



You may notice that it is a bit of overkill to use an entire 4-KB block for these bitmaps; such a bitmap can track whether 32K objects are allocated, and yet we only have 80 inodes and 56 data blocks. However, we just use an entire 4-KB block for each of these bitmaps for simplicity.

The careful reader (i.e., the reader who is still awake) may have noticed there is one block left in the design of the on-disk structure of our very simple file system. We reserve this for the **superblock**, denoted by an S in the diagram below. The superblock contains information about this particular file system, including, for example, how many inodes and data blocks are in the file system (80 and 56, respectively in this instance), where the inode table begins (block 3), and so forth. It will likely also include a magic number of some kind to identify the file system type (in this case, vsfs).



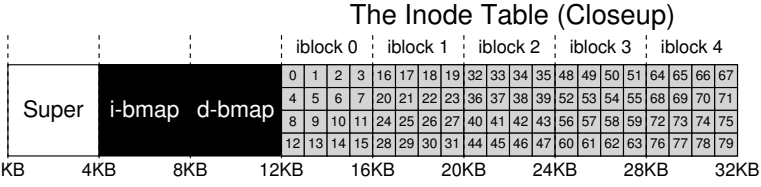
Thus, when mounting a file system, the operating system will read the superblock first, to initialize various parameters, and then attach the volume to the file-system tree. When files within the volume are accessed, the system will thus know exactly where to look for the needed on-disk structures.

40.3 File Organization: The Inode

One of the most important on-disk structures of a file system is the **inode**; virtually all file systems have a structure similar to this. The name inode is short for **index node**, the historical name given to it in UNIX [RT74] and possibly earlier systems, used because these nodes were originally arranged in an array, and the array *indexed* into when accessing a particular inode.

ASIDE: **DATA STRUCTURE — THE INODE**
The **inode** is the generic name that is used in many file systems to describe the structure that holds the metadata for a given file, such as its length, permissions, and the location of its constituent blocks. The name goes back at least as far as UNIX (and probably further back to Multics if not earlier systems); it is short for **index node**, as the inode number is used to index into an array of on-disk inodes in order to find the inode of that number. As we'll see, design of the inode is one key part of file system design. Most modern systems have some kind of structure like this for every file they track, but perhaps call them different things (such as dnodes, fnodes, etc.).

Each inode is implicitly referred to by a number (called the **inumber**), which we've earlier called the **low-level name** of the file. In vsfs (and other simple file systems), given an i-number, you should directly be able to calculate where on the disk the corresponding inode is located. For example, take the inode table of vsfs as above: 20-KB in size (5 4-KB blocks) and thus consisting of 80 inodes (assuming each inode is 256 bytes); further assume that the inode region starts at 12KB (i.e, the superblock starts at 0KB, the inode bitmap is at address 4KB, the data bitmap at 8KB, and thus the inode table comes right after). In vsfs, we thus have the following layout for the beginning of the file system partition (in closeup view):



To read inode number 32, the file system would first calculate the offset into the inode region ($32 \cdot sizeof(inode)$ or 8192), add it to the start address of the inode table on disk (`inodeStartAddr` = $12KB$), and thus arrive upon the correct byte address of the desired block of inodes: $20KB$. Recall that disks are not byte addressable, but rather consist of a large number of addressable sectors, usually 512 bytes. Thus, to fetch the block of inodes that contains inode 32, the file system would issue a read to sector $\frac{20 \times 1024}{512}$, or 40, to fetch the desired inode block. More generally, the sector address `iaddr` of the inode block can be calculated as follows:

```
blk    = (inumber * sizeof(inode_t)) / blockSize;
sector = ((blk * blockSize) + inodeStartAddr) / sectorSize;
```

Inside each inode is virtually all of the information you need about a file: its *type* (e.g., regular file, directory, etc.), its *size*, the number of *blocks*

Size	Name	What is this inode field for?
2	mode	can this file be read/written/executed?
2	uid	who owns this file?
4	size	how many bytes are in this file?
4	time	what time was this file last accessed?
4	ctime	what time was this file created?
4	mtime	what time was this file last modified?
4	dtime	what time was this inode deleted?
2	gid	which group does this file belong to?
2	links_count	how many hard links are there to this file?
4	blocks	how many blocks have been allocated to this file?
4	flags	how should ext2 use this inode?
4	osd1	an OS-dependent field
60	block	a set of disk pointers (15 total)
4	generation	file version (used by NFS)
4	file_acl	a new permissions model beyond mode bits
4	dir_acl	called access control lists

Figure 40.1: **Simplified Ext2 Inode**

allocated to it, *protection information* (such as who owns the file, as well as who can access it), some *time* information, including when the file was created, modified, or last accessed, as well as information about where its data blocks reside on disk (e.g., pointers of some kind). We refer to all such information about a file as **metadata**; in fact, any information inside the file system that isn't pure user data is often referred to as such. An example inode from ext2 [P09] is shown in Figure 40.1[1].

One of the most important decisions in the design of the inode is how it refers to where data blocks are. One simple approach would be to have one or more **direct pointers** (disk addresses) inside the inode; each pointer refers to one disk block that belongs to the file. Such an approach is limited: for example, if you want to have a file that is really big (e.g., bigger than the size of a block multiplied by the number of direct pointers), you are out of luck.

The Multi-Level Index

To support bigger files, file system designers have had to introduce different structures within inodes. One common idea is to have a special pointer known as an **indirect pointer**. Instead of pointing to a block that contains user data, it points to a block that contains more pointers, each of which point to user data. Thus, an inode may have some fixed number of direct pointers (e.g., 12), and a single indirect pointer. If a file grows large enough, an indirect block is allocated (from the data-block region of the disk), and the inode's slot for an indirect pointer is set to point to it. Assuming 4-KB blocks and 4-byte disk addresses, that adds another 1024 pointers; the file can grow to be $(12 + 1024) \cdot 4K$ or 4144KB.

[1] Type info is kept in the directory entry, and thus is not found in the inode itself.

TIP: CONSIDER EXTENT-BASED APPROACHES

A different approach is to use **extents** instead of pointers. An extent is simply a disk pointer plus a length (in blocks); thus, instead of requiring a pointer for every block of a file, all one needs is a pointer and a length to specify the on-disk location of a file. Just a single extent is limiting, as one may have trouble finding a contiguous chunk of on-disk free space when allocating a file. Thus, extent-based file systems often allow for more than one extent, thus giving more freedom to the file system during file allocation.

In comparing the two approaches, pointer-based approaches are the most flexible but use a large amount of metadata per file (particularly for large files). Extent-based approaches are less flexible but more compact; in particular, they work well when there is enough free space on the disk and files can be laid out contiguously (which is the goal for virtually any file allocation policy anyhow).

Not surprisingly, in such an approach, you might want to support even larger files. To do so, just add another pointer to the inode: the **double indirect pointer**. This pointer refers to a block that contains pointers to indirect blocks, each of which contain pointers to data blocks. A double indirect block thus adds the possibility to grow files with an additional $1024 \cdot 1024$ or 1-million 4KB blocks, in other words supporting files that are over 4GB in size. You may want even more, though, and we bet you know where this is headed: the **triple indirect pointer**.

Overall, this imbalanced tree is referred to as the **multi-level index** approach to pointing to file blocks. Let's examine an example with twelve direct pointers, as well as both a single and a double indirect block. Assuming a block size of 4 KB, and 4-byte pointers, this structure can accommodate a file of just over 4 GB in size (i.e., $(12 + 1024 + 1024^2) \times 4\ KB$). Can you figure out how big of a file can be handled with the addition of a triple-indirect block? (hint: pretty big)

Many file systems use a multi-level index, including commonly-used file systems such as Linux ext2 [P09] and ext3, NetApp's WAFL, as well as the original UNIX file system. Other file systems, including SGI XFS and Linux ext4, use **extents** instead of simple pointers; see the earlier aside for details on how extent-based schemes work (they are akin to segments in the discussion of virtual memory).

You might be wondering: why use an imbalanced tree like this? Why not a different approach? Well, as it turns out, many researchers have studied file systems and how they are used, and virtually every time they find certain "truths" that hold across the decades. One such finding is that *most files are small*. This imbalanced design reflects such a reality; if most files are indeed small, it makes sense to optimize for this case. Thus, with a small number of direct pointers (12 is a typical number), an inode

ASIDE: **LINKED-BASED APPROACHES**

Another simpler approach in designing inodes is to use a **linked list**. Thus, inside an inode, instead of having multiple pointers, you just need one, to point to the first block of the file. To handle larger files, add another pointer at the end of that data block, and so on, and thus you can support large files.

As you might have guessed, linked file allocation performs poorly for some workloads; think about reading the last block of a file, for example, or just doing random access. Thus, to make linked allocation work better, some systems will keep an in-memory table of link information, instead of storing the next pointers with the data blocks themselves. The table is indexed by the address of a data block D; the content of an entry is simply D's next pointer, i.e., the address of the next block in a file which follows D. A null-value could be there too (indicating an end-of-file), or some other marker to indicate that a particular block is free. Having such a table of next pointers makes it so that a linked allocation scheme can effectively do random file accesses, simply by first scanning through the (in memory) table to find the desired block, and then accessing (on disk) it directly.

Does such a table sound familiar? What we have described is the basic structure of what is known as the **file allocation table**, or **FAT** file system. Yes, this classic old Windows file system, before NTFS [C94], is based on a simple linked-based allocation scheme. There are other differences from a standard UNIX file system too; for example, there are no inodes per se, but rather directory entries which store metadata about a file and refer directly to the first block of said file, which makes creating hard links impossible. See Brouwer [B02] for more of the inelegant details.

can directly point to 48 KB of data, needing one (or more) indirect blocks for larger files. See Agrawal et. al [A+07] for a recent study; Figure 40.2 summarizes those results.

Of course, in the space of inode design, many other possibilities exist; after all, the inode is just a data structure, and any data structure that stores the relevant information, and can query it effectively, is sufficient. As file system software is readily changed, you should be willing to explore different designs should workloads or technologies change.

Most files are small	Roughly 2K is the most common size
Average file size is growing	Almost 200K is the average
Most bytes are stored in large files	A few big files use most of the space
File systems contains lots of files	Almost 100K on average
File systems are roughly half full	Even as disks grow, file systems remain ~50% full
Directories are typically small	Many have few entries; most have 20 or fewer

Figure 40.2: **File System Measurement Summary**

40.4 Directory Organization

In vsfs (as in many file systems), directories have a simple organization; a directory basically just contains a list of (entry name, inode number) pairs. For each file or directory in a given directory, there is a string and a number in the data block(s) of the directory. For each string, there may also be a length (assuming variable-sized names).

For example, assume a directory `dir` (inode number 5) has three files in it (`foo`, `bar`, and `foobar_is_a_pretty_longname`), with inode numbers 12, 13, and 24 respectively. The on-disk data for `dir` might look like:

```
inum | reclen | strlen | name
   5      12        2     .
   2      12        3     ..
  12      12        4     foo
  13      12        4     bar
  24      36       28     foobar_is_a_pretty_longname
```

In this example, each entry has an inode number, record length (the total bytes for the name plus any left over space), string length (the actual length of the name), and finally the name of the entry. Note that each directory has two extra entries, `.` "dot" and `..` "dot-dot"; the dot directory is just the current directory (in this example, `dir`), whereas dot-dot is the parent directory (in this case, the root).

Deleting a file (e.g., calling `unlink()`) can leave an empty space in the middle of the directory, and hence there should be some way to mark that as well (e.g., with a reserved inode number such as zero). Such a delete is one reason the record length is used: a new entry may reuse an old, bigger entry and thus have extra space within.

You might be wondering where exactly directories are stored. Often, file systems treat directories as a special type of file. Thus, a directory has an inode, somewhere in the inode table (with the type field of the inode marked as "directory" instead of "regular file"). The directory has data blocks pointed to by the inode (and perhaps, indirect blocks); these data blocks live in the data block region of our simple file system. Our on-disk structure thus remains unchanged.

We should also note again that this simple linear list of directory entries is not the only way to store such information. As before, any data structure is possible. For example, XFS [S+96] stores directories in B-tree form, making file create operations (which have to ensure that a file name has not been used before creating it) faster than systems with simple lists that must be scanned in their entirety.

40.5 Free Space Management

A file system must track which inodes and data blocks are free, and which are not, so that when a new file or directory is allocated, it can find space for it. Thus **free space management** is important for all file systems. In vsfs, we have two simple bitmaps for this task.

ASIDE: **FREE SPACE MANAGEMENT**

There are many ways to manage free space; bitmaps are just one way. Some early file systems used **free lists**, where a single pointer in the super block was kept to point to the first free block; inside that block the next free pointer was kept, thus forming a list through the free blocks of the system. When a block was needed, the head block was used and the list updated accordingly.

Modern file systems use more sophisticated data structures. For example, SGI's XFS [S+96] uses some form of a **B-tree** to compactly represent which chunks of the disk are free. As with any data structure, different time-space trade-offs are possible.

For example, when we create a file, we will have to allocate an inode for that file. The file system will thus search through the bitmap for an inode that is free, and allocate it to the file; the file system will have to mark the inode as used (with a 1) and eventually update the on-disk bitmap with the correct information. A similar set of activities take place when a data block is allocated.

Some other considerations might also come into play when allocating data blocks for a new file. For example, some Linux file systems, such as ext2 and ext3, will look for a sequence of blocks (say 8) that are free when a new file is created and needs data blocks; by finding such a sequence of free blocks, and then allocating them to the newly-created file, the file system guarantees that a portion of the file will be contiguous on the disk, thus improving performance. Such a **pre-allocation** policy is thus a commonly-used heuristic when allocating space for data blocks.

40.6 Access Paths: Reading and Writing

Now that we have some idea of how files and directories are stored on disk, we should be able to follow the flow of operation during the activity of reading or writing a file. Understanding what happens on this **access path** is thus the second key in developing an understanding of how a file system works; pay attention!

For the following examples, let us assume that the file system has been mounted and thus that the superblock is already in memory. Everything else (i.e., inodes, directories) is still on the disk.

Reading A File From Disk

In this simple example, let us first assume that you want to simply open a file (e.g., `/foo/bar`), read it, and then close it. For this simple example, let's assume the file is just 4KB in size (i.e., 1 block).

When you issue an `open("/foo/bar", O_RDONLY)` call, the file system first needs to find the inode for the file `bar`, to obtain some basic information about the file (permissions information, file size, etc.). To do so,

	data bitmap	inode bitmap	root inode	foo inode	bar inode	root data	foo data	bar data[0]	bar data[1]	bar data[2]
open(bar)			**read**			**read**				
				read			**read**			
					read					
read()					**read**			**read**		
					write					
read()					**read**				**read**	
					write					
read()					**read**					**read**
					write					

Figure 40.3: **File Read Timeline (Time Increasing Downward)**

the file system must be able to find the inode, but all it has right now is the full pathname. The file system must **traverse** the pathname and thus locate the desired inode.

All traversals begin at the root of the file system, in the **root directory** which is simply called `/`. Thus, the first thing the FS will read from disk is the inode of the root directory. But where is this inode? To find an inode, we must know its i-number. Usually, we find the i-number of a file or directory in its parent directory; the root has no parent (by definition). Thus, the root inode number must be "well known"; the FS must know what it is when the file system is mounted. In most UNIX file systems, the root inode number is 2. Thus, to begin the process, the FS reads in the block that contains inode number 2 (the first inode block).

Once the inode is read in, the FS can look inside of it to find pointers to data blocks, which contain the contents of the root directory. The FS will thus use these on-disk pointers to read through the directory, in this case looking for an entry for `foo`. By reading in one or more directory data blocks, it will find the entry for foo; once found, the FS will also have found the inode number of `foo` (say it is 44) which it will need next.

The next step is to recursively traverse the pathname until the desired inode is found. In this example, the FS reads the block containing the inode of `foo` and then its directory data, finally finding the inode number of `bar`. The final step of `open()` is to read `bar`'s inode into memory; the FS then does a final permissions check, allocates a file descriptor for this process in the per-process open-file table, and returns it to the user.

Once open, the program can then issue a `read()` system call to read from the file. The first read (at offset 0 unless `lseek()` has been called) will thus read in the first block of the file, consulting the inode to find the location of such a block; it may also update the inode with a new last-accessed time. The read will further update the in-memory open file table for this file descriptor, updating the file offset such that the next read will read the second file block, etc.

ASIDE: **READS DON'T ACCESS ALLOCATION STRUCTURES**
We've seen many students get confused by allocation structures such as bitmaps. In particular, many often think that when you are simply reading a file, and not allocating any new blocks, that the bitmap will still be consulted. This is not true! Allocation structures, such as bitmaps, are only accessed when allocation is needed. The inodes, directories, and indirect blocks have all the information they need to complete a read request; there is no need to make sure a block is allocated when the inode already points to it.

At some point, the file will be closed. There is much less work to be done here; clearly, the file descriptor should be deallocated, but for now, that is all the FS really needs to do. No disk I/Os take place.

A depiction of this entire process is found in Figure 40.3 (time increases downward). In the figure, the open causes numerous reads to take place in order to finally locate the inode of the file. Afterwards, reading each block requires the file system to first consult the inode, then read the block, and then update the inode's last-accessed-time field with a write. Spend some time and try to understand what is going on.

Also note that the amount of I/O generated by the open is proportional to the length of the pathname. For each additional directory in the path, we have to read its inode as well as its data. Making this worse would be the presence of large directories; here, we only have to read one block to get the contents of a directory, whereas with a large directory, we might have to read many data blocks to find the desired entry. Yes, life can get pretty bad when reading a file; as you're about to find out, writing out a file (and especially, creating a new one) is even worse.

Writing to Disk

Writing to a file is a similar process. First, the file must be opened (as above). Then, the application can issue `write()` calls to update the file with new contents. Finally, the file is closed.

Unlike reading, writing to the file may also **allocate** a block (unless the block is being overwritten, for example). When writing out a new file, each write not only has to write data to disk but has to first decide which block to allocate to the file and thus update other structures of the disk accordingly (e.g., the data bitmap and inode). Thus, each write to a file logically generates five I/Os: one to read the data bitmap (which is then updated to mark the newly-allocated block as used), one to write the bitmap (to reflect its new state to disk), two more to read and then write the inode (which is updated with the new block's location), and finally one to write the actual block itself.

The amount of write traffic is even worse when one considers a simple and common operation such as file creation. To create a file, the file system must not only allocate an inode, but also allocate space within

	data bitmap	inode bitmap	root inode	foo inode	bar inode	root data	foo data	bar data[0]	bar data[1]	bar data[2]
create (/foo/bar)		**read** **write**	**read**	**read**	**read** **write** **write**	**read**	**read** **write**			
write()	**read** **write**				**read** **write**			**write**		
write()	**read** **write**				**read** **write**				**write**	
write()	**read** **write**				**read** **write**					**write**

Figure 40.4: **File Creation Timeline (Time Increasing Downward)**

the directory containing the new file. The total amount of I/O traffic to do so is quite high: one read to the inode bitmap (to find a free inode), one write to the inode bitmap (to mark it allocated), one write to the new inode itself (to initialize it), one to the data of the directory (to link the high-level name of the file to its inode number), and one read and write to the directory inode to update it. If the directory needs to grow to accommodate the new entry, additional I/Os (i.e., to the data bitmap, and the new directory block) will be needed too. All that just to create a file!

Let's look at a specific example, where the file `/foo/bar` is created, and three blocks are written to it. Figure 40.4 shows what happens during the `open()` (which creates the file) and during each of three 4KB writes.

In the figure, reads and writes to the disk are grouped under which system call caused them to occur, and the rough ordering they might take place in goes from top to bottom of the figure. You can see how much work it is to create the file: 10 I/Os in this case, to walk the pathname and then finally create the file. You can also see that each allocating write costs 5 I/Os: a pair to read and update the inode, another pair to read and update the data bitmap, and then finally the write of the data itself. How can a file system accomplish any of this with reasonable efficiency?

THE CRUX: HOW TO REDUCE FILE SYSTEM I/O COSTS
Even the simplest of operations like opening, reading, or writing a file incurs a huge number of I/O operations, scattered over the disk. What can a file system do to reduce the high costs of doing so many I/Os?

40.7 Caching and Buffering

As the examples above show, reading and writing files can be expensive, incurring many I/Os to the (slow) disk. To remedy what would clearly be a huge performance problem, most file systems aggressively use system memory (DRAM) to cache important blocks.

Imagine the open example above: without caching, every file open would require at least two reads for every level in the directory hierarchy (one to read the inode of the directory in question, and at least one to read its data). With a long pathname (e.g., `/1/2/3/ ... /100/file.txt`), the file system would literally perform hundreds of reads just to open the file!

Early file systems thus introduced a **fixed-size cache** to hold popular blocks. As in our discussion of virtual memory, strategies such as **LRU** and different variants would decide which blocks to keep in cache. This fixed-size cache would usually be allocated at boot time to be roughly 10% of total memory.

This **static partitioning** of memory, however, can be wasteful; what if the file system doesn't need 10% of memory at a given point in time? With the fixed-size approach described above, unused pages in the file cache cannot be re-purposed for some other use, and thus go to waste.

Modern systems, in contrast, employ a **dynamic partitioning** approach. Specifically, many modern operating systems integrate virtual memory pages and file system pages into a **unified page cache** [S00]. In this way, memory can be allocated more flexibly across virtual memory and file system, depending on which needs more memory at a given time.

Now imagine the file open example with caching. The first open may generate a lot of I/O traffic to read in directory inode and data, but subsequent file opens of that same file (or files in the same directory) will mostly hit in the cache and thus no I/O is needed.

Let us also consider the effect of caching on writes. Whereas read I/O can be avoided altogether with a sufficiently large cache, write traffic has to go to disk in order to become persistent. Thus, a cache does not serve as the same kind of filter on write traffic that it does for reads. That said, **write buffering** (as it is sometimes called) certainly has a number of performance benefits. First, by delaying writes, the file system can **batch** some updates into a smaller set of I/Os; for example, if an inode bitmap is updated when one file is created and then updated moments later as another file is created, the file system saves an I/O by delaying the write after the first update. Second, by buffering a number of writes in memory,

TIP: UNDERSTAND STATIC VS. DYNAMIC PARTITIONING

When dividing a resource among different clients/users, you can use either **static partitioning** or **dynamic partitioning**. The static approach simply divides the resource into fixed proportions once; for example, if there are two possible users of memory, you can give some fixed fraction of memory to one user, and the rest to the other. The dynamic approach is more flexible, giving out differing amounts of the resource over time; for example, one user may get a higher percentage of disk bandwidth for a period of time, but then later, the system may switch and decide to give a different user a larger fraction of available disk bandwidth.

Each approach has its advantages. Static partitioning ensures each user receives some share of the resource, usually delivers more predictable performance, and is often easier to implement. Dynamic partitioning can achieve better utilization (by letting resource-hungry users consume otherwise idle resources), but can be more complex to implement, and can lead to worse performance for users whose idle resources get consumed by others and then take a long time to reclaim when needed. As is often the case, there is no best method; rather, you should think about the problem at hand and decide which approach is most suitable. Indeed, shouldn't you always be doing that?

the system can then **schedule** the subsequent I/Os and thus increase performance. Finally, some writes are avoided altogether by delaying them; for example, if an application creates a file and then deletes it, delaying the writes to reflect the file creation to disk **avoids** them entirely. In this case, laziness (in writing blocks to disk) is a virtue.

For the reasons above, most modern file systems buffer writes in memory for anywhere between five and thirty seconds, representing yet another trade-off: if the system crashes before the updates have been propagated to disk, the updates are lost; however, by keeping writes in memory longer, performance can be improved by batching, scheduling, and even avoiding writes.

Some applications (such as databases) don't enjoy this trade-off. Thus, to avoid unexpected data loss due to write buffering, they simply force writes to disk, by calling `fsync()`, by using **direct I/O** interfaces that work around the cache, or by using the **raw disk** interface and avoiding the file system altogether[2]. While most applications live with the trade-offs made by the file system, there are enough controls in place to get the system to do what you want it to, should the default not be satisfying.

[2]Take a database class to learn more about old-school databases and their former insistence on avoiding the OS and controlling everything themselves. But watch out! Those database types are always trying to bad mouth the OS. Shame on you, database people. Shame.

TIP: UNDERSTAND THE DURABILITY/PERFORMANCE TRADE-OFF
Storage systems often present a durability/performance trade-off to users. If the user wishes data that is written to be immediately durable, the system must go through the full effort of committing the newly-written data to disk, and thus the write is slow (but safe). However, if the user can tolerate the loss of a little data, the system can buffer writes in memory for some time and write them later to the disk (in the background). Doing so makes writes appear to complete quickly, thus improving perceived performance; however, if a crash occurs, writes not yet committed to disk will be lost, and hence the trade-off. To understand how to make this trade-off properly, it is best to understand what the application using the storage system requires; for example, while it may be tolerable to lose the last few images downloaded by your web browser, losing part of a database transaction that is adding money to your bank account may be less tolerable. Unless you're rich, of course; in that case, why do you care so much about hoarding every last penny?

40.8 Summary

We have seen the basic machinery required in building a file system. There needs to be some information about each file (metadata), usually stored in a structure called an inode. Directories are just a specific type of file that store name→inode-number mappings. And other structures are needed too; for example, file systems often use a structure such as a bitmap to track which inodes or data blocks are free or allocated.

The terrific aspect of file system design is its freedom; the file systems we explore in the coming chapters each take advantage of this freedom to optimize some aspect of the file system. There are also clearly many policy decisions we have left unexplored. For example, when a new file is created, where should it be placed on disk? This policy and others will also be the subject of future chapters. Or will they?[3]

[3]Cue mysterious music that gets you even more intrigued about the topic of file systems.

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An excellent recent analysis of how file systems are actually used. Use the bibliography within to follow the trail of file-system analysis papers back to the early 1980s.

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Jeff Bonwick and Bill Moore
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A short book about NTFS; there are probably ones with more technical details elsewhere.

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John H. Howard, Michael L. Kazar, Sherri G. Menees, David A. Nichols, M. Satyanarayanan, Robert N. Sidebotham, Michael J. West.
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A classic distributed file system; we'll be learning more about it later, don't worry.

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Some details on ext2, a very simple Linux file system based on FFS, the Berkeley Fast File System. We'll be reading about it in the next chapter.

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M. Ritchie and K. Thompson
CACM, Volume 17:7, pages 365-375, 1974
The original paper about UNIX. Read it to see the underpinnings of much of modern operating systems.

[S00] "UBC: An Efficient Unified I/O and Memory Caching Subsystem for NetBSD"
Chuck Silvers
FREENIX, 2000
A nice paper about NetBSD's integration of file-system buffer caching and the virtual-memory page cache. Many other systems do the same type of thing.

[S+96] "Scalability in the XFS File System"
Adan Sweeney, Doug Doucette, Wei Hu, Curtis Anderson,
Mike Nishimoto, Geoff Peck
USENIX '96, January 1996, San Diego, CA
The first attempt to make scalability of operations, including things like having millions of files in a directory, a central focus. A great example of pushing an idea to the extreme. The key idea behind this file system: everything is a tree. We should have a chapter on this file system too.

Homework

Use this tool, `vsfs.py`, to study how file system state changes as various operations take place. The file system begins in an empty state, with just a root directory. As the simulation takes place, various operations are performed, thus slowly changing the on-disk state of the file system. See the README for details.

Questions

1. Run the simulator with some different random seeds (say 17, 18, 19, 20), and see if you can figure out which operations must have taken place between each state change.
2. Now do the same, using different random seeds (say 21, 22, 23, 24), except run with the `-r` flag, thus making you guess the state change while being shown the operation. What can you conclude about the inode and data-block allocation algorithms, in terms of which blocks they prefer to allocate?
3. Now reduce the number of data blocks in the file system, to very low numbers (say two), and run the simulator for a hundred or so requests. What types of files end up in the file system in this highly-constrained layout? What types of operations would fail?
4. Now do the same, but with inodes. With very few inodes, what types of operations can succeed? Which will usually fail? What is the final state of the file system likely to be?

41

Locality and The Fast File System

When the UNIX operating system was first introduced, the UNIX wizard himself Ken Thompson wrote the first file system. Let's call that the "old UNIX file system", and it was really simple. Basically, its data structures looked like this on the disk:

The super block (S) contained information about the entire file system: how big the volume is, how many inodes there are, a pointer to the head of a free list of blocks, and so forth. The inode region of the disk contained all the inodes for the file system. Finally, most of the disk was taken up by data blocks.

The good thing about the old file system was that it was simple, and supported the basic abstractions the file system was trying to deliver: files and the directory hierarchy. This easy-to-use system was a real step forward from the clumsy, record-based storage systems of the past, and the directory hierarchy was a true advance over simpler, one-level hierarchies provided by earlier systems.

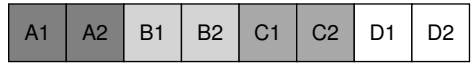
41.1 The Problem: Poor Performance

The problem: performance was terrible. As measured by Kirk McKusick and his colleagues at Berkeley [MJLF84], performance started off bad and got worse over time, to the point where the file system was delivering only 2% of overall disk bandwidth!

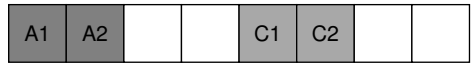
The main issue was that the old UNIX file system treated the disk like it was a random-access memory; data was spread all over the place without regard to the fact that the medium holding the data was a disk, and thus had real and expensive positioning costs. For example, the data blocks of a file were often very far away from its inode, thus inducing an expensive seek whenever one first read the inode and then the data blocks of a file (a pretty common operation).

Worse, the file system would end up getting quite **fragmented**, as the free space was not carefully managed. The free list would end up pointing to a bunch of blocks spread across the disk, and as files got allocated, they would simply take the next free block. The result was that a logically contiguous file would be accessed by going back and forth across the disk, thus reducing performance dramatically.

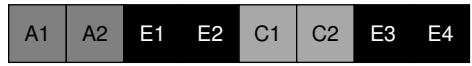
For example, imagine the following data block region, which contains four files (A, B, C, and D), each of size 2 blocks:



If B and D are deleted, the resulting layout is:



As you can see, the free space is fragmented into two chunks of two blocks, instead of one nice contiguous chunk of four. Let's say you now wish to allocate a file E, of size four blocks:



You can see what happens: E gets spread across the disk, and as a result, when accessing E, you don't get peak (sequential) performance from the disk. Rather, you first read E1 and E2, then seek, then read E3 and E4. This fragmentation problem happened all the time in the old UNIX file system, and it hurt performance. A side note: this problem is exactly what disk **defragmentation** tools help with; they reorganize on-disk data to place files contiguously and make free space for one or a few contiguous regions, moving data around and then rewriting inodes and such to reflect the changes.

One other problem: the original block size was too small (512 bytes). Thus, transferring data from the disk was inherently inefficient. Smaller blocks were good because they minimized **internal fragmentation** (waste within the block), but bad for transfer as each block might require a positioning overhead to reach it. Thus, the problem:

THE CRUX:
HOW TO ORGANIZE ON-DISK DATA TO IMPROVE PERFORMANCE

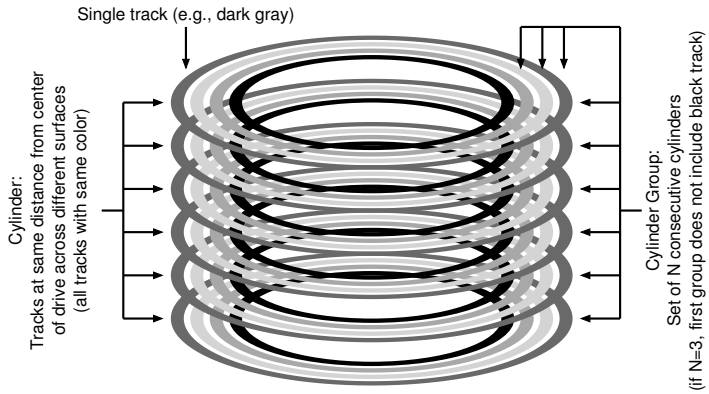
How can we organize file system data structures so as to improve performance? What types of allocation policies do we need on top of those data structures? How do we make the file system "disk aware"?

41.2 FFS: Disk Awareness Is The Solution

A group at Berkeley decided to build a better, faster file system, which they cleverly called the **Fast File System (FFS)**. The idea was to design the file system structures and allocation policies to be "disk aware" and thus improve performance, which is exactly what they did. FFS thus ushered in a new era of file system research; by keeping the same *interface* to the file system (the same APIs, including `open()`, `read()`, `write()`, `close()`, and other file system calls) but changing the internal *implementation*, the authors paved the path for new file system construction, work that continues today. Virtually all modern file systems adhere to the existing interface (and thus preserve compatibility with applications) while changing their internals for performance, reliability, or other reasons.

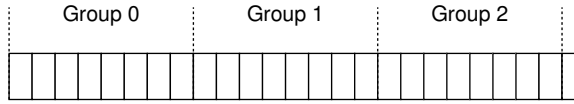
41.3 Organizing Structure: The Cylinder Group

The first step was to change the on-disk structures. FFS divides the disk into a number of **cylinder groups**. A single **cylinder** is a set of tracks on different surfaces of a hard drive that are the same distance from the center of the drive; it is called a cylinder because of its clear resemblance to the so-called geometrical shape. FFS aggregates each N consecutive cylinders into group, and thus the entire disk can thus be viewed as a collection of cylinder groups. Here is a simple example, showing the four outer most tracks of a drive with six platters, and a cylinder group that consists of three cylinders:



Note that modern drives do not export enough information for the file system to truly understand whether a particular cylinder is in use; as discussed previously [AD14a], disks export a logical address space of blocks and hide details of their geometry from clients. Thus, modern file

systems (such as Linux ext2, ext3, and ext4) instead organize the drive into **block groups**, each of which is just a consecutive portion of the disk's address space. The picture below illustrates an example where every 8 blocks are organized into a different block group (note that real groups would consist of many more blocks):



Whether you call them cylinder groups or block groups, these groups are the central mechanism that FFS uses to improve performance. Critically, by placing two files within the same group, FFS can ensure that accessing one after the other will not result in long seeks across the disk.

To use these groups to store files and directories, FFS needs to have the ability to place files and directories into a group, and track all necessary information about them therein. To do so, FFS includes all the structures you might expect a file system to have within each group, e.g., space for inodes, data blocks, and some structures to track whether each of those are allocated or free. Here is a depiction of what FFS keeps within a single cylinder group:

Let's now examine the components of this single cylinder group in more detail. FFS keeps a copy of the **super block** (S) in each group for reliability reasons. The super block is needed to mount the file system; by keeping multiple copies, if one copy becomes corrupt, you can still mount and access the file system by using a working replica.

Within each group, FFS needs to track whether the inodes and data blocks of the group are allocated. A per-group **inode bitmap** (ib) and **data bitmap** (db) serve this role for inodes and data blocks in each group. Bitmaps are an excellent way to manage free space in a file system because it is easy to find a large chunk of free space and allocate it to a file, perhaps avoiding some of the fragmentation problems of the free list in the old file system.

Finally, the **inode** and **data block** regions are just like those in the previous very-simple file system (VSFS). Most of each cylinder group, as usual, is comprised of data blocks.

ASIDE: **FFS FILE CREATION**

As an example, think about what data structures must be updated when a file is created; assume, for this example, that the user creates a new file `/foo/bar.txt` and that the file is one block long (4KB). The file is new, and thus needs a new inode; thus, both the inode bitmap and the newly-allocated inode will be written to disk. The file also has data in it and thus it too must be allocated; the data bitmap and a data block will thus (eventually) be written to disk. Hence, at least four writes to the current cylinder group will take place (recall that these writes may be buffered in memory for a while before they take place). But this is not all! In particular, when creating a new file, you must also place the file in the file-system hierarchy, i.e., the directory must be updated. Specifically, the parent directory `foo` must be updated to add the entry for `bar.txt`; this update may fit in an existing data block of `foo` or require a new block to be allocated (with associated data bitmap). The inode of `foo` must also be updated, both to reflect the new length of the directory as well as to update time fields (such as last-modified-time). Overall, it is a lot of work just to create a new file! Perhaps next time you do so, you should be more thankful, or at least surprised that it all works so well.

41.4 Policies: How To Allocate Files and Directories

With this group structure in place, FFS now has to decide how to place files and directories and associated metadata on disk to improve performance. The basic mantra is simple: *keep related stuff together* (and its corollary, *keep unrelated stuff far apart*).

Thus, to obey the mantra, FFS has to decide what is "related" and place it within the same block group; conversely, unrelated items should be placed into different block groups. To achieve this end, FFS makes use of a few simple placement heuristics.

The first is the placement of directories. FFS employs a simple approach: find the cylinder group with a low number of allocated directories (to balance directories across groups) and a high number of free inodes (to subsequently be able to allocate a bunch of files), and put the directory data and inode in that group. Of course, other heuristics could be used here (e.g., taking into account the number of free data blocks).

For files, FFS does two things. First, it makes sure (in the general case) to allocate the data blocks of a file in the same group as its inode, thus preventing long seeks between inode and data (as in the old file system). Second, it places all files that are in the same directory in the cylinder group of the directory they are in. Thus, if a user creates four files, `/a/b`, `/a/c`, `/a/d`, and `b/f`, FFS would try to place the first three near one another (same group) and the fourth far away (in some other group).

Let's look at an example of such an allocation. In the example, assume that there are only 10 inodes and 10 data blocks in each group (both

unrealistically small numbers), and that the three directories (the root directory `/`, `/a`, and `/b`) and four files (`/a/c`, `/a/d`, `/a/e`, `/b/f`) are placed within them per the FFS policies. Assume the regular files are each two blocks in size, and that the directories have just a single block of data. For this figure, we use the obvious symbols for each file or directory (i.e., `/` for the root directory, `a` for `/a`, `f` for `/b/f`, and so forth).

```
group inodes     data
    0 /--------- /---------
    1 acde------ accddee---
    2 bf-------- bff-------
    3 ---------- ----------
    4 ---------- ----------
    5 ---------- ----------
    6 ---------- ----------
    7 ---------- ----------
    ...
```

Note that the FFS policy does two positive things: the data blocks of each file are near each file's inode, and files in the same directory are near one another (namely, `/a/c`, `/a/d`, and `/a/e` are all in Group 1, and directory `/b` and its file `/b/f` are near one another in Group 2).

In contrast, let's now look at an inode allocation policy that simply spreads inodes across groups, trying to ensure that no group's inode table fills up quickly. The final allocation might thus look something like this:

```
group inodes     data
    0 /--------- /---------
    1 a--------- a---------
    2 b--------- b---------
    3 c--------- cc--------
    4 d--------- dd--------
    5 e--------- ee--------
    6 f--------- ff--------
    7 ---------- ----------
    ...
```

As you can see from the figure, while this policy does indeed keep file (and directory) data near its respective inode, files within a directory are arbitrarily spread around the disk, and thus name-based locality is not preserved. Access to files `/a/c`, `/a/d`, and `/a/e` now spans three groups instead of one as per the FFS approach.

The FFS policy heuristics are not based on extensive studies of file-system traffic or anything particularly nuanced; rather, they are based on good old-fashioned **common sense** (isn't that what CS stands for after all?)[1]. Files in a directory *are* often accessed together: imagine compiling a bunch of files and then linking them into a single executable. Because such namespace-based locality exists, FFS will often improve performance, making sure that seeks between related files are nice and short.

[1]Some people refer to common sense as **horse sense**, especially people who work regularly with horses. However, we have a feeling that this idiom may be lost as the "mechanized horse", a.k.a. the car, gains in popularity. What will they invent next? A flying machine??!!

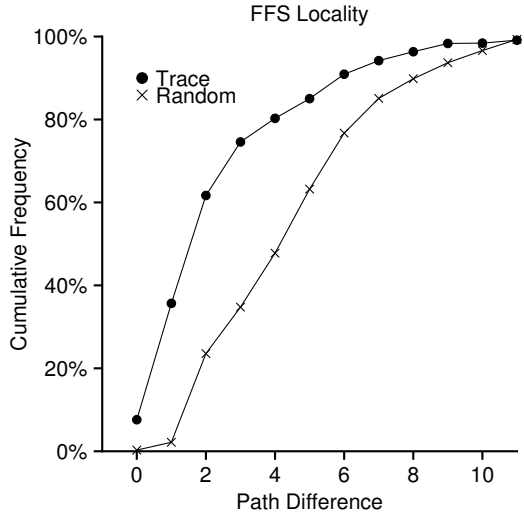


Figure 41.1: **FFS Locality For SEER Traces**

41.5 Measuring File Locality

To understand better whether these heuristics make sense, let's analyze some traces of file system access and see if indeed there is namespace locality. For some reason, there doesn't seem to be a good study of this topic in the literature.

Specifically, we'll use the SEER traces [K94] and analyze how "far away" file accesses were from one another in the directory tree. For example, if file `f` is opened, and then re-opened next in the trace (before any other files are opened), the distance between these two opens in the directory tree is zero (as they are the same file). If a file `f` in directory `dir` (i.e., `dir/f`) is opened, and followed by an open of file `g` in the same directory (i.e., `dir/g`), the distance between the two file accesses is one, as they share the same directory but are not the same file. Our distance metric, in other words, measures how far up the directory tree you have to travel to find the *common ancestor* of two files; the closer they are in the tree, the lower the metric.

Figure 41.1 shows the locality observed in the SEER traces over all workstations in the SEER cluster over the entirety of all traces. The graph plots the difference metric along the x-axis, and shows the cumulative percentage of file opens that were of that difference along the y-axis. Specifically, for the SEER traces (marked "Trace" in the graph), you can see that about 7% of file accesses were to the file that was opened previously, and that nearly 40% of file accesses were to either the same file or to one in the same directory (i.e., a difference of zero or one). Thus, the FFS locality assumption seems to make sense (at least for these traces).

Interestingly, another 25% or so of file accesses were to files that had a distance of two. This type of locality occurs when the user has structured a set of related directories in a multi-level fashion and consistently jumps between them. For example, if a user has a `src` directory and builds object files (`.o` files) into an `obj` directory, and both of these directories are sub-directories of a main `proj` directory, a common access pattern will be `proj/src/foo.c` followed by `proj/obj/foo.o`. The distance between these two accesses is two, as `proj` is the common ancestor. FFS does *not* capture this type of locality in its policies, and thus more seeking will occur between such accesses.

For comparison, the graph also shows locality for a "Random" trace. The random trace was generated by selecting files from within an existing SEER trace in random order, and calculating the distance metric between these randomly-ordered accesses. As you can see, there is less namespace locality in the random traces, as expected. However, because eventually every file shares a common ancestor (e.g., the root), there is some locality, and thus random is useful as a comparison point.

41.6 The Large-File Exception

In FFS, there is one important exception to the general policy of file placement, and it arises for large files. Without a different rule, a large file would entirely fill the block group it is first placed within (and maybe others). Filling a block group in this manner is undesirable, as it prevents subsequent "related" files from being placed within this block group, and thus may hurt file-access locality.

Thus, for large files, FFS does the following. After some number of blocks are allocated into the first block group (e.g., 12 blocks, or the number of direct pointers available within an inode), FFS places the next "large" chunk of the file (e.g., those pointed to by the first indirect block) in another block group (perhaps chosen for its low utilization). Then, the next chunk of the file is placed in yet another different block group, and so on.

Let's look at some diagrams to understand this policy better. Without the large-file exception, a single large file would place all of its blocks into one part of the disk. We investigate a small example of a file (`/a`) with 30 blocks in an FFS configured with 10 inodes and 40 data blocks per group. Here is the depiction of FFS without the large-file exception:

```
group inodes     data
    0 /a-------- /aaaaaaaaa aaaaaaaaaa aaaaaaaaaa a---------
    1 ---------- ---------- ---------- ---------- ----------
    2 ---------- ---------- ---------- ---------- ----------
    ...
```

As you can see in the picture, `/a` fills up most of the data blocks in Group 0, whereas other groups remain empty. If some other files are now created in the root directory (`/`), there is not much room for their data in the group.

With the large-file exception (here set to five blocks in each chunk), FFS instead spreads the file spread across groups, and the resulting utilization within any one group is not too high:

```
group inodes     data
    0 /a-------- /aaaaa---- ---------- ---------- ----------
    1 ---------- aaaaa----- ---------- ---------- ----------
    2 ---------- aaaaa----- ---------- ---------- ----------
    3 ---------- aaaaa----- ---------- ---------- ----------
    4 ---------- aaaaa----- ---------- ---------- ----------
    5 ---------- aaaaa----- ---------- ---------- ----------
    6 ---------- ---------- ---------- ---------- ----------
    ...
```

The astute reader (that's you) will note that spreading blocks of a file across the disk will hurt performance, particularly in the relatively common case of sequential file access (e.g., when a user or application reads chunks 0 through 29 in order). And you are right, oh astute reader of ours! But you can address this problem by choosing chunk size carefully.

Specifically, if the chunk size is large enough, the file system will spend most of its time transferring data from disk and just a (relatively) little time seeking between chunks of the block. This process of reducing an overhead by doing more work per overhead paid is called **amortization** and is a common technique in computer systems.

Let's do an example: assume that the average positioning time (i.e., seek and rotation) for a disk is 10 ms. Assume further that the disk transfers data at 40 MB/s. If your goal was to spend half our time seeking between chunks and half our time transferring data (and thus achieve 50% of peak disk performance), you would thus need to spend 10 ms transferring data for every 10 ms positioning. So the question becomes: how big does a chunk have to be in order to spend 10 ms in transfer? Easy, just use our old friend, math, in particular the dimensional analysis mentioned in the chapter on disks [AD14a]:

$$\frac{40\ \cancel{MB}}{\cancel{sec}} \cdot \frac{1024\ KB}{1\ \cancel{MB}} \cdot \frac{1\ \cancel{sec}}{1000\ \cancel{ms}} \cdot 10\ \cancel{ms} = 409.6\ KB \tag{41.1}$$

Basically, what this equation says is this: if you transfer data at 40 MB/s, you need to transfer only 409.6KB every time you seek in order to spend half your time seeking and half your time transferring. Similarly, you can compute the size of the chunk you would need to achieve 90% of peak bandwidth (turns out it is about 3.69MB), or even 99% of peak bandwidth (40.6MB!). As you can see, the closer you want to get to peak, the bigger these chunks get (see Figure 41.2 for a plot of these values).

FFS did not use this type of calculation in order to spread large files across groups, however. Instead, it took a simple approach, based on the structure of the inode itself. The first twelve direct blocks were placed in the same group as the inode; each subsequent indirect block, and all the blocks it pointed to, was placed in a different group. With a block size of 4KB, and 32-bit disk addresses, this strategy implies that every

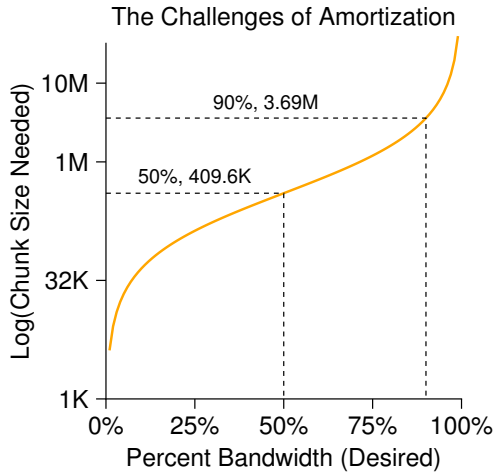


Figure 41.2: **Amortization: How Big Do Chunks Have To Be?**

1024 blocks of the file (4MB) were placed in separate groups, the lone exception being the first 48KB of the file as pointed to by direct pointers.

Note that the trend in disk drives is that transfer rate improves fairly rapidly, as disk manufacturers are good at cramming more bits into the same surface, but the mechanical aspects of drives related to seeks (disk arm speed and the rate of rotation) improve rather slowly [P98]. The implication is that over time, mechanical costs become relatively more expensive, and thus, to amortize said costs, you have to transfer more data between seeks.

41.7 A Few Other Things About FFS

FFS introduced a few other innovations too. In particular, the designers were extremely worried about accommodating small files; as it turned out, many files were 2KB or so in size back then, and using 4KB blocks, while good for transferring data, was not so good for space efficiency. This **internal fragmentation** could thus lead to roughly half the disk being wasted for a typical file system.

The solution the FFS designers hit upon was simple and solved the problem. They decided to introduce **sub-blocks**, which were 512-byte little blocks that the file system could allocate to files. Thus, if you created a small file (say 1KB in size), it would occupy two sub-blocks and thus not waste an entire 4KB block. As the file grew, the file system will continue allocating 512-byte blocks to it until it acquires a full 4KB of data. At that point, FFS will find a 4KB block, *copy* the sub-blocks into it, and free the sub-blocks for future use.

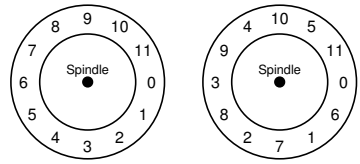


Figure 41.3: **FFS: Standard Versus Parameterized Placement**

You might observe that this process is inefficient, requiring a lot of extra work for the file system (in particular, a lot of extra I/O to perform the copy). And you'd be right again! Thus, FFS generally avoided this pessimal behavior by modifying the `libc` library; the library would buffer writes and then issue them in 4KB chunks to the file system, thus avoiding the sub-block specialization entirely in most cases.

A second neat thing that FFS introduced was a disk layout that was optimized for performance. In those times (before SCSI and other more modern device interfaces), disks were much less sophisticated and required the host CPU to control their operation in a more hands-on way. A problem arose in FFS when a file was placed on consecutive sectors of the disk, as on the left in Figure 41.3.

In particular, the problem arose during sequential reads. FFS would first issue a read to block 0; by the time the read was complete, and FFS issued a read to block 1, it was too late: block 1 had rotated under the head and now the read to block 1 would incur a full rotation.

FFS solved this problem with a different layout, as you can see on the right in Figure 41.3. By skipping over every other block (in the example), FFS has enough time to request the next block before it went past the disk head. In fact, FFS was smart enough to figure out for a particular disk *how many* blocks it should skip in doing layout in order to avoid the extra rotations; this technique was called **parameterization**, as FFS would figure out the specific performance parameters of the disk and use those to decide on the exact staggered layout scheme.

You might be thinking: this scheme isn't so great after all. In fact, you will only get 50% of peak bandwidth with this type of layout, because you have to go around each track twice just to read each block once. Fortunately, modern disks are much smarter: they internally read the entire track in and buffer it in an internal disk cache (often called a **track buffer** for this very reason). Then, on subsequent reads to the track, the disk will just return the desired data from its cache. File systems thus no longer have to worry about these incredibly low-level details. Abstraction and higher-level interfaces can be a good thing, when designed properly.

Some other usability improvements were added as well. FFS was one of the first file systems to allow for **long file names**, thus enabling more expressive names in the file system instead of the traditional fixed-size approach (e.g., 8 characters). Further, a new concept was introduced

TIP: MAKE THE SYSTEM USABLE
Probably the most basic lesson from FFS is that not only did it introduce the conceptually good idea of disk-aware layout, but it also added a number of features that simply made the system more usable. Long file names, symbolic links, and a rename operation that worked atomically all improved the utility of a system; while hard to write a research paper about (imagine trying to read a 14-pager about "The Symbolic Link: Hard Link's Long Lost Cousin"), such small features made FFS more useful and thus likely increased its chances for adoption. Making a system usable is often as or more important than its deep technical innovations.

called a **symbolic link**. As discussed in a previous chapter [AD14b] , hard links are limited in that they both could not point to directories (for fear of introducing loops in the file system hierarchy) and that they can only point to files within the same volume (i.e., the inode number must still be meaningful). Symbolic links allow the user to create an "alias" to any other file or directory on a system and thus are much more flexible. FFS also introduced an atomic `rename()` operation for renaming files. Usability improvements, beyond the basic technology, also likely gained FFS a stronger user base.

41.8 Summary

The introduction of FFS was a watershed moment in file system history, as it made clear that the problem of file management was one of the most interesting issues within an operating system, and showed how one might begin to deal with that most important of devices, the hard disk. Since that time, hundreds of new file systems have developed, but still today many file systems take cues from FFS (e.g., Linux ext2 and ext3 are obvious intellectual descendants). Certainly all modern systems account for the main lesson of FFS: treat the disk like it's a disk.

References

[AD14a] "Operating Systems: Three Easy Pieces"
Chapter: Hard Disk Drives
Remzi Arpaci-Dusseau and Andrea Arpaci-Dusseau
There is no way you should be reading about FFS without having first understood hard drives in some detail. If you try to do so, please instead go directly to jail; do not pass go, and, critically, do not collect 200 much-needed simoleons.

[AD14b] "Operating Systems: Three Easy Pieces"
Chapter: File System Implementation
Remzi Arpaci-Dusseau and Andrea Arpaci-Dusseau
As above, it makes little sense to read this chapter unless you have read (and understood) the chapter on file system implementation. Otherwise, we'll be throwing around terms like "inode" and "indirect block" and you'll be like "huh?" and that is no fun for either of us.

[K94] "The Design of the SEER Predictive Caching System"
G. H. Kuenning
MOBICOMM '94, Santa Cruz, California, December 1994
According to Kuenning, this is the best overview of the SEER project, which led to (among other things) the collection of these traces.

[MJLF84] "A Fast File System for UNIX"
Marshall K. McKusick, William N. Joy, Sam J. Leffler, Robert S. Fabry
ACM Transactions on Computing Systems, 2:3, pages 181-197.
August, 1984. *McKusick was recently honored with the IEEE Reynold B. Johnson award for his contributions to file systems, much of which was based on his work building FFS. In his acceptance speech, he discussed the original FFS software: only 1200 lines of code! Modern versions are a little more complex, e.g., the BSD FFS descendant now is in the 50-thousand lines-of-code range.*

[P98] "Hardware Technology Trends and Database Opportunities"
David A. Patterson
Keynote Lecture at the ACM SIGMOD Conference (SIGMOD '98)
June, 1998
A great and simple overview of disk technology trends and how they change over time.

Homework

This section introduces `ffs.py`, a simple FFS simulator you can use to understand better how FFS-based file and directory allocation work. See the README for details on how to run the simulator.

Questions

1. Examine the file `in.largefile`, and then run the simulator with flag `-f in.largefile` and `-L 4`. The latter sets the large file exception to 4 blocks in a group before moving on to the next one. What do you think the file system allocation will look like? Then run with `-c` enabled to see the actual layout.
2. Now run with `-L 30`. What do you expect to see? Once again, turn on `-c` to see if you were right. You can also use `-S` to see exactly which blocks were allocated to the file `/a`.
3. Now we will compute some statistics about the file. The first is something we call *filespan*, which is the max distance between any two data blocks of the file or between the inode and any data block. Calculate the filespan of `/a`. Run `ffs.py -f in.largefile -L 4 -T -c` to see what it is. Do the same with `-L 100`. What difference do you expect in filespan as the large-file exception parameter changes from low values to high values?
4. Now let's look at a new input file, `in.manyfiles`. How do you think the FFS policy will lay these files out across groups? (you can run with `-v` to see what files and directories are created, or just `cat in.manyfiles`). Run the simulator with `-c` to see if you were right.
5. A new metric we will use to evaluate FFS is called *dirspan*. This metric calculates the spread of files within a particular directory, specifically the max distance between the inodes and data blocks of all the files in the directory as well as the inode and data block of the directory itself. Run with `in.manyfiles` and the `-T` flag, and see if you can figure out the dirspan of the three directories. Run with `-c` to see if you were right. How good of a job does FFS do in minimizing dirspan?
6. Now change the size of the inode table per group to 5 (`-I 5`). How do you think this will change the layout of the files? Run with `-c` to see if you were right. How does it affect the dirspan?
7. One policy that can affect FFS effectiveness is which group to place the inode of a new directory in. The default policy (in the simulator) simply looks for the group with the most free inodes. A slightly different policy, specified with `-A`, looks for a group of groups with the most free inodes. For example, if you run with `-A 2`, when allocating a new directory, the simulator will look at groups in pairs and pick the best pair for the allocation. Now you should run `./ffs.py -f in.manyfiles -I 5 -A 2 -c` to see how allo-

cation changes with this strategy. How does it affect dirspan? Why might this policy be a good idea?

8. One last policy change we will explore relates to file fragmentation. Run `./ffs.py -f in.fragmented -v` and see if you can predict how the files that remain are allocated. Run with `-c` to confirm your answer. What is interesting about the data layout of file `/i`? Why is it problematic?
9. A new policy, which we call *contiguous allocation* and enabled with the `-C` flag, tries to ensure that each file is allocated contiguously. Specifically, with `-C n`, the file system tries to ensure that `n` contiguous blocks are free within a group before allocating a block. Run `./ffs.py -f in.fragmented -v -C 2 -c` to see the difference in layout. How does layout change as the parameter passed to `-C` increases? Finally, how does `-C` affect filespan and dirspan?

42

Crash Consistency: FSCK and Journaling

As we've seen thus far, the file system manages a set of data structures to implement the expected abstractions: files, directories, and all of the other metadata needed to support the basic abstraction that we expect from a file system. Unlike most data structures (for example, those found in memory of a running program), file system data structures must **persist**, i.e., they must survive over the long haul, stored on devices that retain data despite power loss (such as hard disks or flash-based SSDs).

One major challenge faced by a file system is how to update persistent data structures despite the presence of a **power loss** or **system crash**. Specifically, what happens if, right in the middle of updating on-disk structures, someone trips over the power cord and the machine loses power? Or the operating system encounters a bug and crashes? Because of power losses and crashes, updating a persistent data structure can be quite tricky, and leads to a new and interesting problem in file system implementation, known as the **crash-consistency problem**.

This problem is quite simple to understand. Imagine you have to update two on-disk structures, A and B, in order to complete a particular operation. Because the disk only services a single request at a time, one of these requests will reach the disk first (either A or B). If the system crashes or loses power after one write completes, the on-disk structure will be left in an **inconsistent** state. And thus, we have a problem that all file systems need to solve:

THE CRUX: HOW TO UPDATE THE DISK DESPITE CRASHES

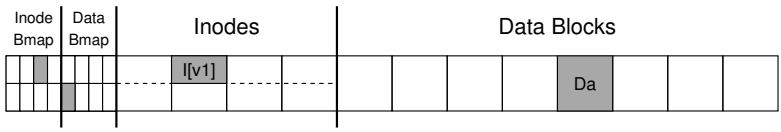
The system may crash or lose power between any two writes, and thus the on-disk state may only partially get updated. After the crash, the system boots and wishes to mount the file system again (in order to access files and such). Given that crashes can occur at arbitrary points in time, how do we ensure the file system keeps the on-disk image in a reasonable state?

In this chapter, we'll describe this problem in more detail, and look at some methods file systems have used to overcome it. We'll begin by examining the approach taken by older file systems, known as **fsck** or the **file system checker**. We'll then turn our attention to another approach, known as **journaling** (also known as **write-ahead logging**), a technique which adds a little bit of overhead to each write but recovers more quickly from crashes or power losses. We will discuss the basic machinery of journaling, including a few different flavors of journaling that Linux ext3 [T98,PAA05] (a relatively modern journaling file system) implements.

42.1 A Detailed Example

To kick off our investigation of journaling, let's look at an example. We'll need to use a **workload** that updates on-disk structures in some way. Assume here that the workload is simple: the append of a single data block to an existing file. The append is accomplished by opening the file, calling `lseek()` to move the file offset to the end of the file, and then issuing a single 4KB write to the file before closing it.

Let's also assume we are using standard simple file system structures on the disk, similar to file systems we have seen before. This tiny example includes an **inode bitmap** (with just 8 bits, one per inode), a **data bitmap** (also 8 bits, one per data block), inodes (8 total, numbered 0 to 7, and spread across four blocks), and data blocks (8 total, numbered 0 to 7). Here is a diagram of this file system:



If you look at the structures in the picture, you can see that a single inode is allocated (inode number 2), which is marked in the inode bitmap, and a single allocated data block (data block 4), also marked in the data bitmap. The inode is denoted I[v1], as it is the first version of this inode; it will soon be updated (due to the workload described above).

Let's peek inside this simplified inode too. Inside of I[v1], we see:

```
owner       : remzi
permissions : read-write
size        : 1
pointer     : 4
pointer     : null
pointer     : null
pointer     : null
```

In this simplified inode, the `size` of the file is `1` (it has one block allocated), the first direct pointer points to block `4` (the first data block of the file, Da), and all three other direct pointers are set to `null` (indicating

that they are not used). Of course, real inodes have many more fields; see previous chapters for more information.

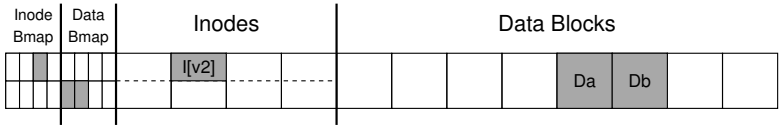
When we append to the file, we are adding a new data block to it, and thus must update three on-disk structures: the inode (which must point to the new block as well as have a bigger size due to the append), the new data block Db, and a new version of the data bitmap (call it B[v2]) to indicate that the new data block has been allocated.

Thus, in the memory of the system, we have three blocks which we must write to disk. The updated inode (inode version 2, or I[v2] for short) now looks like this:

```
owner       : remzi
permissions : read-write
size        : 2
pointer     : 4
pointer     : 5
pointer     : null
pointer     : null
```

The updated data bitmap (B[v2]) now looks like this: 00001100. Finally, there is the data block (Db), which is just filled with whatever it is users put into files. Stolen music perhaps?

What we would like is for the final on-disk image of the file system to look like this:



To achieve this transition, the file system must perform three separate writes to the disk, one each for the inode (I[v2]), bitmap (B[v2]), and data block (Db). Note that these writes usually don't happen immediately when the user issues a `write()` system call; rather, the dirty inode, bitmap, and new data will sit in main memory (in the **page cache** or **buffer cache**) for some time first; then, when the file system finally decides to write them to disk (after say 5 seconds or 30 seconds), the file system will issue the requisite write requests to the disk. Unfortunately, a crash may occur and thus interfere with these updates to the disk. In particular, if a crash happens after one or two of these writes have taken place, but not all three, the file system could be left in a funny state.

Crash Scenarios

To understand the problem better, let's look at some example crash scenarios. Imagine only a single write succeeds; there are thus three possible outcomes, which we list here:

- **Just the data block (Db) is written to disk.** In this case, the data is on disk, but there is no inode that points to it and no bitmap that even says the block is allocated. Thus, it is as if the write never occurred. This case is not a problem at all, from the perspective of file-system crash consistency[1].

- **Just the updated inode (I[v2]) is written to disk.** In this case, the inode points to the disk address (5) where Db was about to be written, but Db has not yet been written there. Thus, if we trust that pointer, we will read **garbage** data from the disk (the old contents of disk address 5).

  Further, we have a new problem, which we call a **file-system inconsistency**. The on-disk bitmap is telling us that data block 5 has not been allocated, but the inode is saying that it has. This disagreement in the file system data structures is an inconsistency in the data structures of the file system; to use the file system, we must somehow resolve this problem (more on that below).

- **Just the updated bitmap (B[v2]) is written to disk.** In this case, the bitmap indicates that block 5 is allocated, but there is no inode that points to it. Thus the file system is inconsistent again; if left unresolved, this write would result in a **space leak**, as block 5 would never be used by the file system.

There are also three more crash scenarios in this attempt to write three blocks to disk. In these cases, two writes succeed and the last one fails:

- **The inode (I[v2]) and bitmap (B[v2]) are written to disk, but not data (Db).** In this case, the file system metadata is completely consistent: the inode has a pointer to block 5, the bitmap indicates that 5 is in use, and thus everything looks OK from the perspective of the file system's metadata. But there is one problem: 5 has garbage in it again.

- **The inode (I[v2]) and the data block (Db) are written, but not the bitmap (B[v2]).** In this case, we have the inode pointing to the correct data on disk, but again have an inconsistency between the inode and the old version of the bitmap (B1). Thus, we once again need to resolve the problem before using the file system.

- **The bitmap (B[v2]) and data block (Db) are written, but not the inode (I[v2]).** In this case, we again have an inconsistency between the inode and the data bitmap. However, even though the block was written and the bitmap indicates its usage, we have no idea which file it belongs to, as no inode points to the file.

[1] However, it might be a problem for the user, who just lost some data!

The Crash Consistency Problem

Hopefully, from these crash scenarios, you can see the many problems that can occur to our on-disk file system image because of crashes: we can have inconsistency in file system data structures; we can have space leaks; we can return garbage data to a user; and so forth. What we'd like to do ideally is move the file system from one consistent state (e.g., before the file got appended to) to another **atomically** (e.g., after the inode, bitmap, and new data block have been written to disk). Unfortunately, we can't do this easily because the disk only commits one write at a time, and crashes or power loss may occur between these updates. We call this general problem the **crash-consistency problem** (we could also call it the **consistent-update problem**).

42.2 Solution #1: The File System Checker

Early file systems took a simple approach to crash consistency. Basically, they decided to let inconsistencies happen and then fix them later (when rebooting). A classic example of this lazy approach is found in a tool that does this: **fsck**[2]. `fsck` is a UNIX tool for finding such inconsistencies and repairing them [M86]; similar tools to check and repair a disk partition exist on different systems. Note that such an approach can't fix all problems; consider, for example, the case above where the file system looks consistent but the inode points to garbage data. The only real goal is to make sure the file system metadata is internally consistent.

The tool `fsck` operates in a number of phases, as summarized in McKusick and Kowalski's paper [MK96]. It is run *before* the file system is mounted and made available (`fsck` assumes that no other file-system activity is on-going while it runs); once finished, the on-disk file system should be consistent and thus can be made accessible to users.

Here is a basic summary of what `fsck` does:

- **Superblock:** `fsck` first checks if the superblock looks reasonable, mostly doing sanity checks such as making sure the file system size is greater than the number of blocks allocated. Usually the goal of these sanity checks is to find a suspect (corrupt) superblock; in this case, the system (or administrator) may decide to use an alternate copy of the superblock.
- **Free blocks:** Next, `fsck` scans the inodes, indirect blocks, double indirect blocks, etc., to build an understanding of which blocks are currently allocated within the file system. It uses this knowledge to produce a correct version of the allocation bitmaps; thus, if there is any inconsistency between bitmaps and inodes, it is resolved by trusting the information within the inodes. The same type of check is performed for all the inodes, making sure that all inodes that look like they are in use are marked as such in the inode bitmaps.

[2] Pronounced either "eff-ess-see-kay", "eff-ess-check", or, if you don't like the tool, "eff-suck". Yes, serious professional people use this term.

- **Inode state:** Each inode is checked for corruption or other problems. For example, `fsck` makes sure that each allocated inode has a valid type field (e.g., regular file, directory, symbolic link, etc.). If there are problems with the inode fields that are not easily fixed, the inode is considered suspect and cleared by `fsck`; the inode bitmap is correspondingly updated.
- **Inode links:** `fsck` also verifies the link count of each allocated inode. As you may recall, the link count indicates the number of different directories that contain a reference (i.e., a link) to this particular file. To verify the link count, `fsck` scans through the entire directory tree, starting at the root directory, and builds its own link counts for every file and directory in the file system. If there is a mismatch between the newly-calculated count and that found within an inode, corrective action must be taken, usually by fixing the count within the inode. If an allocated inode is discovered but no directory refers to it, it is moved to the `lost+found` directory.
- **Duplicates:** `fsck` also checks for duplicate pointers, i.e., cases where two different inodes refer to the same block. If one inode is obviously bad, it may be cleared. Alternately, the pointed-to block could be copied, thus giving each inode its own copy as desired.
- **Bad blocks:** A check for bad block pointers is also performed while scanning through the list of all pointers. A pointer is considered "bad" if it obviously points to something outside its valid range, e.g., it has an address that refers to a block greater than the partition size. In this case, `fsck` can't do anything too intelligent; it just removes (clears) the pointer from the inode or indirect block.
- **Directory checks:** `fsck` does not understand the contents of user files; however, directories hold specifically formatted information created by the file system itself. Thus, `fsck` performs additional integrity checks on the contents of each directory, making sure that "." and ".." are the first entries, that each inode referred to in a directory entry is allocated, and ensuring that no directory is linked to more than once in the entire hierarchy.

As you can see, building a working `fsck` requires intricate knowledge of the file system; making sure such a piece of code works correctly in all cases can be challenging [G+08]. However, fsck (and similar approaches) have a bigger and perhaps more fundamental problem: they are *too slow*. With a very large disk volume, scanning the entire disk to find all the allocated blocks and read the entire directory tree may take many minutes or hours. Performance of `fsck`, as disks grew in capacity and RAIDs grew in popularity, became prohibitive (despite recent advances [M+13]).

At a higher level, the basic premise of `fsck` seems just a tad irrational. Consider our example above, where just three blocks are written to the disk; it is incredibly expensive to scan the entire disk to fix problems that occurred during an update of just three blocks. This situation is akin to dropping your keys on the floor in your bedroom, and then com-

mencing a *search-the-entire-house-for-keys* recovery algorithm, starting in the basement and working your way through every room. It works but is wasteful. Thus, as disks (and RAIDs) grew, researchers and practitioners started to look for other solutions.

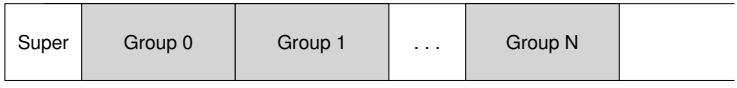
42.3 Solution #2: Journaling (or Write-Ahead Logging)

Probably the most popular solution to the consistent update problem is to steal an idea from the world of database management systems. That idea, known as **write-ahead logging**, was invented to address exactly this type of problem. In file systems, we usually call write-ahead logging **journaling** for historical reasons. The first file system to do this was Cedar [H87], though many modern file systems use the idea, including Linux ext3 and ext4, reiserfs, IBM's JFS, SGI's XFS, and Windows NTFS.

The basic idea is as follows. When updating the disk, before overwriting the structures in place, first write down a little note (somewhere else on the disk, in a well-known location) describing what you are about to do. Writing this note is the "write ahead" part, and we write it to a structure that we organize as a "log"; hence, write-ahead logging.

By writing the note to disk, you are guaranteeing that if a crash takes places during the update (overwrite) of the structures you are updating, you can go back and look at the note you made and try again; thus, you will know exactly what to fix (and how to fix it) after a crash, instead of having to scan the entire disk. By design, journaling thus adds a bit of work during updates to greatly reduce the amount of work required during recovery.

We'll now describe how **Linux ext3**, a popular journaling file system, incorporates journaling into the file system. Most of the on-disk structures are identical to **Linux ext2**, e.g., the disk is divided into block groups, and each block group has an inode and data bitmap as well as inodes and data blocks. The new key structure is the journal itself, which occupies some small amount of space within the partition or on another device. Thus, an ext2 file system (without journaling) looks like this:



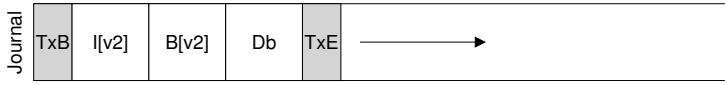
Assuming the journal is placed within the same file system image (though sometimes it is placed on a separate device, or as a file within the file system), an ext3 file system with a journal looks like this:

The real difference is just the presence of the journal, and of course, how it is used.

Data Journaling

Let's look at a simple example to understand how **data journaling** works. Data journaling is available as a mode with the Linux ext3 file system, from which much of this discussion is based.

Say we have our canonical update again, where we wish to write the inode (I[v2]), bitmap (B[v2]), and data block (Db) to disk again. Before writing them to their final disk locations, we are now first going to write them to the log (a.k.a. journal). This is what this will look like in the log:



You can see we have written five blocks here. The transaction begin (TxB) tells us about this update, including information about the pending update to the file system (e.g., the final addresses of the blocks I[v2], B[v2], and Db), as well as some kind of **transaction identifier** (**TID**). The middle three blocks just contain the exact contents of the blocks themselves; this is known as **physical logging** as we are putting the exact physical contents of the update in the journal (an alternate idea, **logical logging**, puts a more compact logical representation of the update in the journal, e.g., "this update wishes to append data block Db to file X", which is a little more complex but can save space in the log and perhaps improve performance). The final block (TxE) is a marker of the end of this transaction, and will also contain the TID.

Once this transaction is safely on disk, we are ready to overwrite the old structures in the file system; this process is called **checkpointing**. Thus, to **checkpoint** the file system (i.e., bring it up to date with the pending update in the journal), we issue the writes I[v2], B[v2], and Db to their disk locations as seen above; if these writes complete successfully, we have successfully checkpointed the file system and are basically done. Thus, our initial sequence of operations:

1. **Journal write:** Write the transaction, including a transaction-begin block, all pending data and metadata updates, and a transaction-end block, to the log; wait for these writes to complete.
2. **Checkpoint:** Write the pending metadata and data updates to their final locations in the file system.

In our example, we would write TxB, I[v2], B[v2], Db, and TxE to the journal first. When these writes complete, we would complete the update by checkpointing I[v2], B[v2], and Db, to their final locations on disk.

Things get a little trickier when a crash occurs during the writes to the journal. Here, we are trying to write the set of blocks in the transaction (e.g., TxB, I[v2], B[v2], Db, TxE) to disk. One simple way to do this would be to issue each one at a time, waiting for each to complete, and then issuing the next. However, this is slow. Ideally, we'd like to issue

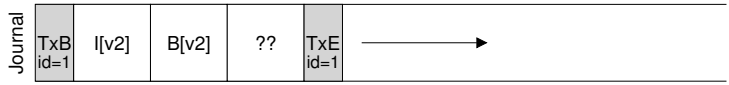
ASIDE: **FORCING WRITES TO DISK**

To enforce ordering between two disk writes, modern file systems have to take a few extra precautions. In olden times, forcing ordering between two writes, A and B, was easy: just issue the write of A to the disk, wait for the disk to interrupt the OS when the write is complete, and then issue the write of B.

Things got slightly more complex due to the increased use of write caches within disks. With write buffering enabled (sometimes called **immediate reporting**), a disk will inform the OS the write is complete when it simply has been placed in the disk's memory cache, and has not yet reached disk. If the OS then issues a subsequent write, it is not guaranteed to reach the disk after previous writes; thus ordering between writes is not preserved. One solution is to disable write buffering. However, more modern systems take extra precautions and issue explicit **write barriers**; such a barrier, when it completes, guarantees that all writes issued before the barrier will reach disk before any writes issued after the barrier.

All of this machinery requires a great deal of trust in the correct operation of the disk. Unfortunately, recent research shows that some disk manufacturers, in an effort to deliver "higher performing" disks, explicitly ignore write-barrier requests, thus making the disks seemingly run faster but at the risk of incorrect operation [C+13, R+11]. As Kahan said, the fast almost always beats out the slow, even if the fast is wrong.

all five block writes at once, as this would turn five writes into a single sequential write and thus be faster. However, this is unsafe, for the following reason: given such a big write, the disk internally may perform scheduling and complete small pieces of the big write in any order. Thus, the disk internally may (1) write TxB, I[v2], B[v2], and TxE and only later (2) write Db. Unfortunately, if the disk loses power between (1) and (2), this is what ends up on disk:



Why is this a problem? Well, the transaction looks like a valid transaction (it has a begin and an end with matching sequence numbers). Further, the file system can't look at that fourth block and know it is wrong; after all, it is arbitrary user data. Thus, if the system now reboots and runs recovery, it will replay this transaction, and ignorantly copy the contents of the garbage block '??' to the location where Db is supposed to live. This is bad for arbitrary user data in a file; it is much worse if it happens to a critical piece of file system, such as the superblock, which could render the file system unmountable.

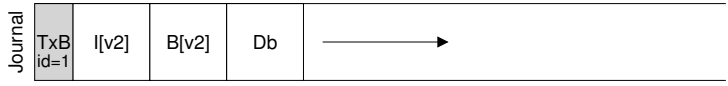
ASIDE: **OPTIMIZING LOG WRITES**

You may have noticed a particular inefficiency of writing to the log. Namely, the file system first has to write out the transaction-begin block and contents of the transaction; only after these writes complete can the file system send the transaction-end block to disk. The performance impact is clear, if you think about how a disk works: usually an extra rotation is incurred (think about why).

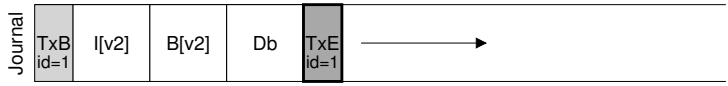
One of our former graduate students, Vijayan Prabhakaran, had a simple idea to fix this problem [P+05]. When writing a transaction to the journal, include a checksum of the contents of the journal in the begin and end blocks. Doing so enables the file system to write the entire transaction at once, without incurring a wait; if, during recovery, the file system sees a mismatch in the computed checksum versus the stored checksum in the transaction, it can conclude that a crash occurred during the write of the transaction and thus discard the file-system update. Thus, with a small tweak in the write protocol and recovery system, a file system can achieve faster common-case performance; on top of that, the system is slightly more reliable, as any reads from the journal are now protected by a checksum.

This simple fix was attractive enough to gain the notice of Linux file system developers, who then incorporated it into the next generation Linux file system, called (you guessed it!) **Linux ext4**. It now ships on millions of machines worldwide, including the Android handheld platform. Thus, every time you write to disk on many Linux-based systems, a little code developed at Wisconsin makes your system a little faster and more reliable.

To avoid this problem, the file system issues the transactional write in two steps. First, it writes all blocks except the TxE block to the journal, issuing these writes all at once. When these writes complete, the journal will look something like this (assuming our append workload again):



When those writes complete, the file system issues the write of the TxE block, thus leaving the journal in this final, safe state:



An important aspect of this process is the atomicity guarantee provided by the disk. It turns out that the disk guarantees that any 512-byte

write will either happen or not (and never be half-written); thus, to make sure the write of TxE is atomic, one should make it a single 512-byte block. Thus, our current protocol to update the file system, with each of its three phases labeled:

1. **Journal write:** Write the contents of the transaction (including TxB, metadata, and data) to the log; wait for these writes to complete.
2. **Journal commit:** Write the transaction commit block (containing TxE) to the log; wait for write to complete; transaction is said to be **committed**.
3. **Checkpoint:** Write the contents of the update (metadata and data) to their final on-disk locations.

Recovery

Let's now understand how a file system can use the contents of the journal to **recover** from a crash. A crash may happen at any time during this sequence of updates. If the crash happens before the transaction is written safely to the log (i.e., before Step 2 above completes), then our job is easy: the pending update is simply skipped. If the crash happens after the transaction has committed to the log, but before the checkpoint is complete, the file system can **recover** the update as follows. When the system boots, the file system recovery process will scan the log and look for transactions that have committed to the disk; these transactions are thus **replayed** (in order), with the file system again attempting to write out the blocks in the transaction to their final on-disk locations. This form of logging is one of the simplest forms there is, and is called **redo logging**. By recovering the committed transactions in the journal, the file system ensures that the on-disk structures are consistent, and thus can proceed by mounting the file system and readying itself for new requests.

Note that it is fine for a crash to happen at any point during checkpointing, even after some of the updates to the final locations of the blocks have completed. In the worst case, some of these updates are simply performed again during recovery. Because recovery is a rare operation (only taking place after an unexpected system crash), a few redundant writes are nothing to worry about[3].

Batching Log Updates

You might have noticed that the basic protocol could add a lot of extra disk traffic. For example, imagine we create two files in a row, called `file1` and `file2`, in the same directory. To create one file, one has to update a number of on-disk structures, minimally including: the inode bitmap (to allocated a new inode), the newly-created inode of the file, the

[3]Unless you worry about everything, in which case we can't help you. Stop worrying so much, it is unhealthy! But now you're probably worried about over-worrying.

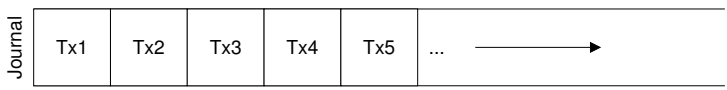
data block of the parent directory containing the new directory entry, as well as the parent directory inode (which now has a new modification time). With journaling, we logically commit all of this information to the journal for each of our two file creations; because the files are in the same directory, and assuming they even have inodes within the same inode block, this means that if we're not careful, we'll end up writing these same blocks over and over.

To remedy this problem, some file systems do not commit each update to disk one at a time (e.g., Linux ext3); rather, one can buffer all updates into a global transaction. In our example above, when the two files are created, the file system just marks the in-memory inode bitmap, inodes of the files, directory data, and directory inode as dirty, and adds them to the list of blocks that form the current transaction. When it is finally time to write these blocks to disk (say, after a timeout of 5 seconds), this single global transaction is committed containing all of the updates described above. Thus, by buffering updates, a file system can avoid excessive write traffic to disk in many cases.

Making The Log Finite

We thus have arrived at a basic protocol for updating file-system on-disk structures. The file system buffers updates in memory for some time; when it is finally time to write to disk, the file system first carefully writes out the details of the transaction to the journal (a.k.a. write-ahead log); after the transaction is complete, the file system checkpoints those blocks to their final locations on disk.

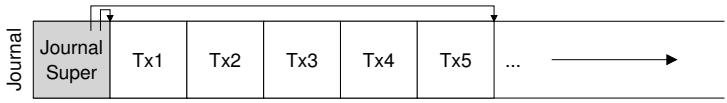
However, the log is of a finite size. If we keep adding transactions to it (as in this figure), it will soon fill. What do you think happens then?



Two problems arise when the log becomes full. The first is simpler, but less critical: the larger the log, the longer recovery will take, as the recovery process must replay all the transactions within the log (in order) to recover. The second is more of an issue: when the log is full (or nearly full), no further transactions can be committed to the disk, thus making the file system "less than useful" (i.e., useless).

To address these problems, journaling file systems treat the log as a circular data structure, re-using it over and over; this is why the journal is sometimes referred to as a **circular log**. To do so, the file system must take action some time after a checkpoint. Specifically, once a transaction has been checkpointed, the file system should free the space it was occupying within the journal, allowing the log space to be reused. There are many ways to achieve this end; for example, you could simply mark the

oldest and newest non-checkpointed transactions in the log in a **journal superblock**; all other space is free. Here is a graphical depiction:



In the journal superblock (not to be confused with the main file system superblock), the journaling system records enough information to know which transactions have not yet been checkpointed, and thus reduces recovery time as well as enables re-use of the log in a circular fashion. And thus we add another step to our basic protocol:

1. **Journal write:** Write the contents of the transaction (containing TxB and the contents of the update) to the log; wait for these writes to complete.
2. **Journal commit:** Write the transaction commit block (containing TxE) to the log; wait for the write to complete; the transaction is now **committed**.
3. **Checkpoint:** Write the contents of the update to their final locations within the file system.
4. **Free:** Some time later, mark the transaction free in the journal by updating the journal superblock.

Thus we have our final data journaling protocol. But there is still a problem: we are writing each data block to the disk *twice*, which is a heavy cost to pay, especially for something as rare as a system crash. Can you figure out a way to retain consistency without writing data twice?

Metadata Journaling

Although recovery is now fast (scanning the journal and replaying a few transactions as opposed to scanning the entire disk), normal operation of the file system is slower than we might desire. In particular, for each write to disk, we are now also writing to the journal first, thus doubling write traffic; this doubling is especially painful during sequential write workloads, which now will proceed at half the peak write bandwidth of the drive. Further, between writes to the journal and writes to the main file system, there is a costly seek, which adds noticeable overhead for some workloads.

Because of the high cost of writing every data block to disk twice, people have tried a few different things in order to speed up performance. For example, the mode of journaling we described above is often called **data journaling** (as in Linux ext3), as it journals all user data (in addition to the metadata of the file system). A simpler (and more common) form of journaling is sometimes called **ordered journaling** (or just **metadata**

journaling), and it is nearly the same, except that user data is *not* written to the journal. Thus, when performing the same update as above, the following information would be written to the journal:

The data block Db, previously written to the log, would instead be written to the file system proper, avoiding the extra write; given that most I/O traffic to the disk is data, not writing data twice substantially reduces the I/O load of journaling. The modification does raise an interesting question, though: when should we write data blocks to disk?

Let's again consider our example append of a file to understand the problem better. The update consists of three blocks: I[v2], B[v2], and Db. The first two are both metadata and will be logged and then checkpointed; the latter will only be written once to the file system. When should we write Db to disk? Does it matter?

As it turns out, the ordering of the data write does matter for metadata-only journaling. For example, what if we write Db to disk *after* the transaction (containing I[v2] and B[v2]) completes? Unfortunately, this approach has a problem: the file system is consistent but I[v2] may end up pointing to garbage data. Specifically, consider the case where I[v2] and B[v2] are written but Db did not make it to disk. The file system will then try to recover. Because Db is *not* in the log, the file system will replay writes to I[v2] and B[v2], and produce a consistent file system (from the perspective of file-system metadata). However, I[v2] will be pointing to garbage data, i.e., at whatever was in the slot where Db was headed.

To ensure this situation does not arise, some file systems (e.g., Linux ext3) write data blocks (of regular files) to the disk *first*, before related metadata is written to disk. Specifically, the protocol is as follows:

1. **Data write:** Write data to final location; wait for completion (the wait is optional; see below for details).
2. **Journal metadata write:** Write the begin block and metadata to the log; wait for writes to complete.
3. **Journal commit:** Write the transaction commit block (containing TxE) to the log; wait for the write to complete; the transaction (including data) is now **committed**.
4. **Checkpoint metadata:** Write the contents of the metadata update to their final locations within the file system.
5. **Free:** Later, mark the transaction free in journal superblock.

By forcing the data write first, a file system can guarantee that a pointer will never point to garbage. Indeed, this rule of "write the pointed-to object before the object that points to it" is at the core of crash consistency, and is exploited even further by other crash consistency schemes [GP94] (see below for details).

In most systems, metadata journaling (akin to ordered journaling of ext3) is more popular than full data journaling. For example, Windows NTFS and SGI's XFS both use some form of metadata journaling. Linux ext3 gives you the option of choosing either data, ordered, or unordered modes (in unordered mode, data can be written at any time). All of these modes keep metadata consistent; they vary in their semantics for data.

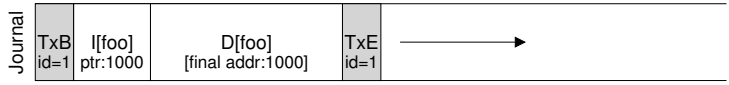
Finally, note that forcing the data write to complete (Step 1) before issuing writes to the journal (Step 2) is not required for correctness, as indicated in the protocol above. Specifically, it would be fine to issue data writes as well as the transaction-begin block and metadata to the journal; the only real requirement is that Steps 1 and 2 complete before the issuing of the journal commit block (Step 3).

Tricky Case: Block Reuse

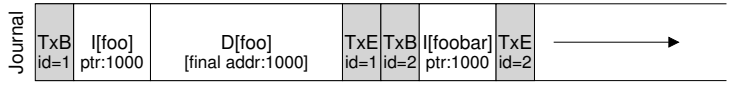
There are some interesting corner cases that make journaling more tricky, and thus are worth discussing. A number of them revolve around block reuse; as Stephen Tweedie (one of the main forces behind ext3) said:

> "What's the hideous part of the entire system? ... It's deleting files. Everything to do with delete is hairy. Everything to do with delete... you have nightmares around what happens if blocks get deleted and then reallocated." [T00]

The particular example Tweedie gives is as follows. Suppose you are using some form of metadata journaling (and thus data blocks for files are *not* journaled). Let's say you have a directory called `foo`. The user adds an entry to `foo` (say by creating a file), and thus the contents of `foo` (because directories are considered metadata) are written to the log; assume the location of the `foo` directory data is block 1000. The log thus contains something like this:



At this point, the user deletes everything in the directory as well as the directory itself, freeing up block 1000 for reuse. Finally, the user creates a new file (say `foobar`), which ends up reusing the same block (1000) that used to belong to `foo`. The inode of `foobar` is committed to disk, as is its data; note, however, because metadata journaling is in use, only the inode of `foobar` is committed to the journal; the newly-written data in block 1000 in the file `foobar` is *not* journaled.



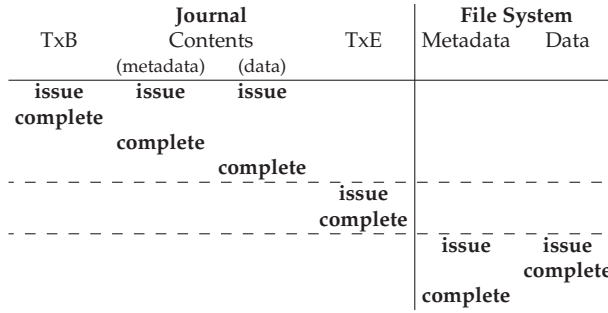


Figure 42.1: **Data Journaling Timeline**

Now assume a crash occurs and all of this information is still in the log. During replay, the recovery process simply replays everything in the log, including the write of directory data in block 1000; the replay thus overwrites the user data of current file `foobar` with old directory contents! Clearly this is not a correct recovery action, and certainly it will be a surprise to the user when reading the file `foobar`.

There are a number of solutions to this problem. One could, for example, never reuse blocks until the delete of said blocks is checkpointed out of the journal. What Linux ext3 does instead is to add a new type of record to the journal, known as a **revoke** record. In the case above, deleting the directory would cause a revoke record to be written to the journal. When replaying the journal, the system first scans for such revoke records; any such revoked data is never replayed, thus avoiding the problem mentioned above.

Wrapping Up Journaling: A Timeline

Before ending our discussion of journaling, we summarize the protocols we have discussed with timelines depicting each of them. Figure 42.1 shows the protocol when journaling data as well as metadata, whereas Figure 42.2 shows the protocol when journaling only metadata.

In each figure, time increases in the downward direction, and each row in the figure shows the logical time that a write can be issued or might complete. For example, in the data journaling protocol (Figure 42.1), the writes of the transaction begin block (TxB) and the contents of the transaction can logically be issued at the same time, and thus can be completed in any order; however, the write to the transaction end block (TxE) must not be issued until said previous writes complete. Similarly, the checkpointing writes to data and metadata blocks cannot begin until the transaction end block has committed. Horizontal dashed lines show where write-ordering requirements must be obeyed.

A similar timeline is shown for the metadata journaling protocol. Note that the data write can logically be issued at the same time as the writes

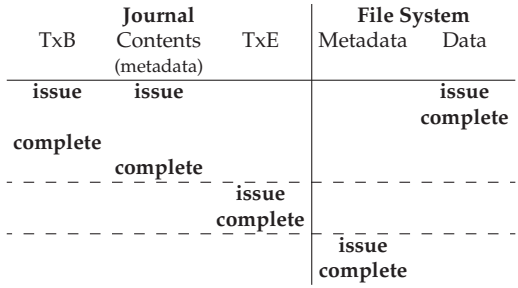


Figure 42.2: **Metadata Journaling Timeline**

to the transaction begin and the contents of the journal; however, it must be issued and complete before the transaction end has been issued.

Finally, note that the time of completion marked for each write in the timelines is arbitrary. In a real system, completion time is determined by the I/O subsystem, which may reorder writes to improve performance. The only guarantees about ordering that we have are those that must be enforced for protocol correctness (and are shown via the horizontal dashed lines in the figures).

42.4 Solution #3: Other Approaches

We've thus far described two options in keeping file system metadata consistent: a lazy approach based on `fsck`, and a more active approach known as journaling. However, these are not the only two approaches. One such approach, known as Soft Updates [GP94], was introduced by Ganger and Patt. This approach carefully orders all writes to the file system to ensure that the on-disk structures are never left in an inconsistent state. For example, by writing a pointed-to data block to disk *before* the inode that points to it, we can ensure that the inode never points to garbage; similar rules can be derived for all the structures of the file system. Implementing Soft Updates can be a challenge, however; whereas the journaling layer described above can be implemented with relatively little knowledge of the exact file system structures, Soft Updates requires intricate knowledge of each file system data structure and thus adds a fair amount of complexity to the system.

Another approach is known as **copy-on-write** (yes, **COW**), and is used in a number of popular file systems, including Sun's ZFS [B07]. This technique never overwrites files or directories in place; rather, it places new updates to previously unused locations on disk. After a number of updates are completed, COW file systems flip the root structure of the file system to include pointers to the newly updated structures. Doing so makes keeping the file system consistent straightforward. We'll be learning more about this technique when we discuss the log-structured file system (LFS) in a future chapter; LFS is an early example of a COW.

Another approach is one we just developed here at Wisconsin. In this technique, entitled **backpointer-based consistency** (or **BBC**), no ordering is enforced between writes. To achieve consistency, an additional **back pointer** is added to every block in the system; for example, each data block has a reference to the inode to which it belongs. When accessing a file, the file system can determine if the file is consistent by checking if the forward pointer (e.g., the address in the inode or direct block) points to a block that refers back to it. If so, everything must have safely reached disk and thus the file is consistent; if not, the file is inconsistent, and an error is returned. By adding back pointers to the file system, a new form of lazy crash consistency can be attained [C+12].

Finally, we also have explored techniques to reduce the number of times a journal protocol has to wait for disk writes to complete. Entitled **optimistic crash consistency** [C+13], this new approach issues as many writes to disk as possible and uses a generalized form of the **transaction checksum** [P+05], as well as a few other techniques, to detect inconsistencies should they arise. For some workloads, these optimistic techniques can improve performance by an order of magnitude. However, to truly function well, a slightly different disk interface is required [C+13].

42.5 Summary

We have introduced the problem of crash consistency, and discussed various approaches to attacking this problem. The older approach of building a file system checker works but is likely too slow to recover on modern systems. Thus, many file systems now use journaling. Journaling reduces recovery time from O(size-of-the-disk-volume) to O(size-of-the-log), thus speeding recovery substantially after a crash and restart. For this reason, many modern file systems use journaling. We have also seen that journaling can come in many different forms; the most commonly used is ordered metadata journaling, which reduces the amount of traffic to the journal while still preserving reasonable consistency guarantees for both file system metadata as well as user data.

References

[B07] "ZFS: The Last Word in File Systems"
Jeff Bonwick and Bill Moore
Available: http://opensolaris.org/os/community/zfs/docs/zfs_last.pdf
ZFS uses copy-on-write and journaling, actually, as in some cases, logging writes to disk will perform better.

[C+12] "Consistency Without Ordering"
Vijay Chidambaram, Tushar Sharma, Andrea C. Arpaci-Dusseau, Remzi H. Arpaci-Dusseau
FAST '12, San Jose, California
A recent paper of ours about a new form of crash consistency based on back pointers. Read it for the exciting details!

[C+13] "Optimistic Crash Consistency"
Vijay Chidambaram, Thanu S. Pillai, Andrea C. Arpaci-Dusseau, Remzi H. Arpaci-Dusseau
SOSP '13, Nemacolin Woodlands Resort, PA, November 2013
Our work on a more optimistic and higher performance journaling protocol. For workloads that call `fsync()` a lot, performance can be greatly improved.

[GP94] "Metadata Update Performance in File Systems"
Gregory R. Ganger and Yale N. Patt
OSDI '94
A clever paper about using careful ordering of writes as the main way to achieve consistency. Implemented later in BSD-based systems.

[G+08] "SQCK: A Declarative File System Checker"
Haryadi S. Gunawi, Abhishek Rajimwale, Andrea C. Arpaci-Dusseau, Remzi H. Arpaci-Dusseau
OSDI '08, San Diego, California
Our own paper on a new and better way to build a file system checker using SQL queries. We also show some problems with the existing checker, finding numerous bugs and odd behaviors, a direct result of the complexity of `fsck`.

[H87] "Reimplementing the Cedar File System Using Logging and Group Commit"
Robert Hagmann
SOSP '87, Austin, Texas, November 1987
The first work (that we know of) that applied write-ahead logging (a.k.a. journaling) to a file system.

[M+13] "ffsck: The Fast File System Checker"
Ao Ma, Chris Dragga, Andrea C. Arpaci-Dusseau, Remzi H. Arpaci-Dusseau
FAST '13, San Jose, California, February 2013
A recent paper of ours detailing how to make fsck an order of magnitude faster. Some of the ideas have already been incorporated into the BSD file system checker [MK96] and are deployed today.

[MK96] "Fsck - The UNIX File System Check Program"
Marshall Kirk McKusick and T. J. Kowalski
Revised in 1996
Describes the first comprehensive file-system checking tool, the eponymous `fsck`. Written by some of the same people who brought you FFS.

[MJLF84] "A Fast File System for UNIX"
Marshall K. McKusick, William N. Joy, Sam J. Leffler, Robert S. Fabry
ACM Transactions on Computing Systems.
August 1984, Volume 2:3
You already know enough about FFS, right? But yeah, it is OK to reference papers like this more than once in a book.

[P+05] "IRON File Systems"
Vijayan Prabhakaran, Lakshmi N. Bairavasundaram, Nitin Agrawal, Haryadi S. Gunawi, Andrea C. Arpaci-Dusseau, Remzi H. Arpaci-Dusseau
SOSP '05, Brighton, England, October 2005
A paper mostly focused on studying how file systems react to disk failures. Towards the end, we introduce a transaction checksum to speed up logging, which was eventually adopted into Linux ext4.

[PAA05] "Analysis and Evolution of Journaling File Systems"
Vijayan Prabhakaran, Andrea C. Arpaci-Dusseau, Remzi H. Arpaci-Dusseau
USENIX '05, Anaheim, California, April 2005
An early paper we wrote analyzing how journaling file systems work.

[R+11] "Coerced Cache Eviction and Discreet-Mode Journaling"
Abhishek Rajimwale, Vijay Chidambaram, Deepak Ramamurthi,
Andrea C. Arpaci-Dusseau, Remzi H. Arpaci-Dusseau
DSN '11, Hong Kong, China, June 2011
Our own paper on the problem of disks that buffer writes in a memory cache instead of forcing them to disk, even when explicitly told not to do that! Our solution to overcome this problem: if you want A to be written to disk before B, first write A, then send a lot of "dummy" writes to disk, hopefully causing A to be forced to disk to make room for them in the cache. A neat if impractical solution.

[T98] "Journaling the Linux ext2fs File System"
Stephen C. Tweedie
The Fourth Annual Linux Expo, May 1998
Tweedie did much of the heavy lifting in adding journaling to the Linux ext2 file system; the result, not surprisingly, is called ext3. Some nice design decisions include the strong focus on backwards compatibility, e.g., you can just add a journaling file to an existing ext2 file system and then mount it as an ext3 file system.

[T00] "EXT3, Journaling Filesystem"
Stephen Tweedie
Talk at the Ottawa Linux Symposium, July 2000
olstrans.sourceforge.net/release/OLS2000-ext3/OLS2000-ext3.html
A transcript of a talk given by Tweedie on ext3.

[T01] "The Linux ext2 File System"
Theodore Ts'o, June, 2001.
Available: http://e2fsprogs.sourceforge.net/ext2.html
A simple Linux file system based on the ideas found in FFS. For a while it was quite heavily used; now it is really just in the kernel as an example of a simple file system.

43

Log-structured File Systems

In the early 90's, a group at Berkeley led by Professor John Ousterhout and graduate student Mendel Rosenblum developed a new file system known as the log-structured file system [RO91]. Their motivation to do so was based on the following observations:

- **System memories are growing**: As memory gets bigger, more data can be cached in memory. As more data is cached, disk traffic increasingly consists of writes, as reads are serviced by the cache. Thus, file system performance is largely determined by its write performance.
- **There is a large gap between random I/O performance and sequential I/O performance**: Hard-drive transfer bandwidth has increased a great deal over the years [P98]; as more bits are packed into the surface of a drive, the bandwidth when accessing said bits increases. Seek and rotational delay costs, however, have decreased slowly; it is challenging to make cheap and small motors spin the platters faster or move the disk arm more quickly. Thus, if you are able to use disks in a sequential manner, you gain a sizeable performance advantage over approaches that cause seeks and rotations.
- **Existing file systems perform poorly on many common workloads**: For example, FFS [MJLF84] would perform a large number of writes to create a new file of size one block: one for a new inode, one to update the inode bitmap, one to the directory data block that the file is in, one to the directory inode to update it, one to the new data block that is a part of the new file, and one to the data bitmap to mark the data block as allocated. Thus, although FFS places all of these blocks within the same block group, FFS incurs many short seeks and subsequent rotational delays and thus performance falls far short of peak sequential bandwidth.
- **File systems are not RAID-aware**: For example, both RAID-4 and RAID-5 have the **small-write problem** where a logical write to a single block causes 4 physical I/Os to take place. Existing file systems do not try to avoid this worst-case RAID writing behavior.

TIP: DETAILS MATTER
All interesting systems are comprised of a few general ideas and a number of details. Sometimes, when you are learning about these systems, you think to yourself "Oh, I get the general idea; the rest is just details," and you use this to only half-learn how things really work. Don't do this! Many times, the details are critical. As we'll see with LFS, the general idea is easy to understand, but to really build a working system, you have to think through *all* of the tricky cases.

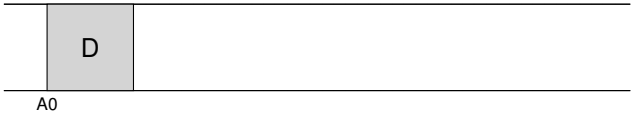
An ideal file system would thus focus on write performance, and try to make use of the sequential bandwidth of the disk. Further, it would perform well on common workloads that not only write out data but also update on-disk metadata structures frequently. Finally, it would work well on RAIDs as well as single disks.

The new type of file system Rosenblum and Ousterhout introduced was called **LFS**, short for the **Log-structured File System**. When writing to disk, LFS first buffers all updates (including metadata!) in an in-memory **segment**; when the segment is full, it is written to disk in one long, sequential transfer to an unused part of the disk. LFS never overwrites existing data, but rather *always* writes segments to free locations. Because segments are large, the disk is used efficiently, and performance of the file system approaches its zenith.

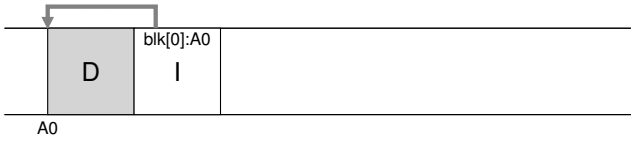
THE CRUX:
HOW TO MAKE ALL WRITES SEQUENTIAL WRITES?
How can a file system transform all writes into sequential writes? For reads, this task is impossible, as the desired block to be read may be anywhere on disk. For writes, however, the file system always has a choice, and it is exactly this choice we hope to exploit.

43.1 Writing To Disk Sequentially

We thus have our first challenge: how do we transform all updates to file-system state into a series of sequential writes to disk? To understand this better, let's use a simple example. Imagine we are writing a data block D to a file. Writing the data block to disk might result in the following on-disk layout, with D written at disk address $A0$:



However, when a user writes a data block, it is not only data that gets written to disk; there is also other **metadata** that needs to be updated. In this case, let's also write the **inode** (I) of the file to disk, and have it point to the data block D. When written to disk, the data block and inode would look something like this (note that the inode looks as big as the data block, which generally isn't the case; in most systems, data blocks are 4 KB in size, whereas an inode is much smaller, around 128 bytes):



This basic idea, of simply writing all updates (such as data blocks, inodes, etc.) to the disk sequentially, sits at the heart of LFS. If you understand this, you get the basic idea. But as with all complicated systems, the devil is in the details.

43.2 Writing Sequentially And Effectively

Unfortunately, writing to disk sequentially is not (alone) enough to guarantee efficient writes. For example, imagine if we wrote a single block to address A, at time T. We then wait a little while, and write to the disk at address $A + 1$ (the next block address in sequential order), but at time $T + \delta$. In-between the first and second writes, unfortunately, the disk has rotated; when you issue the second write, it will thus wait for most of a rotation before being committed (specifically, if the rotation takes time $T_{rotation}$, the disk will wait $T_{rotation} - \delta$ before it can commit the second write to the disk surface). And thus you can hopefully see that simply writing to disk in sequential order is not enough to achieve peak performance; rather, you must issue a large number of *contiguous* writes (or one large write) to the drive in order to achieve good write performance.

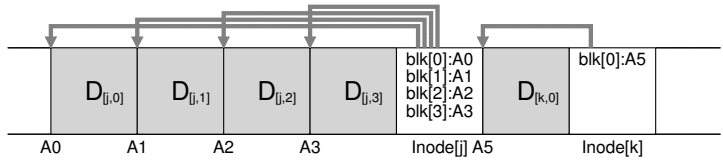
To achieve this end, LFS uses an ancient technique known as **write buffering**[1]. Before writing to the disk, LFS keeps track of updates in memory; when it has received a sufficient number of updates, it writes them to disk all at once, thus ensuring efficient use of the disk.

The large chunk of updates LFS writes at one time is referred to by the name of a **segment**. Although this term is over-used in computer systems, here it just means a large-ish chunk which LFS uses to group writes. Thus, when writing to disk, LFS buffers updates in an in-memory

[1] Indeed, it is hard to find a good citation for this idea, since it was likely invented by many and very early on in the history of computing. For a study of the benefits of write buffering, see Solworth and Orji [SO90]; to learn about its potential harms, see Mogul [M94].

segment, and then writes the segment all at once to the disk. As long as the segment is large enough, these writes will be efficient.

Here is an example, in which LFS buffers two sets of updates into a small segment; actual segments are larger (a few MB). The first update is of four block writes to file j; the second is one block being added to file k. LFS then commits the entire segment of seven blocks to disk at once. The resulting on-disk layout of these blocks is as follows:



43.3 How Much To Buffer?

This raises the following question: how many updates should LFS buffer before writing to disk? The answer, of course, depends on the disk itself, specifically how high the positioning overhead is in comparison to the transfer rate; see the FFS chapter for a similar analysis.

For example, assume that positioning (i.e., rotation and seek overheads) before each write takes roughly $T_{position}$ seconds. Assume further that the disk transfer rate is R_{peak} MB/s. How much should LFS buffer before writing when running on such a disk?

The way to think about this is that every time you write, you pay a fixed overhead of the positioning cost. Thus, how much do you have to write in order to **amortize** that cost? The more you write, the better (obviously), and the closer you get to achieving peak bandwidth.

To obtain a concrete answer, let's assume we are writing out D MB. The time to write out this chunk of data (T_{write}) is the positioning time $T_{position}$ plus the time to transfer D ($\frac{D}{R_{peak}}$), or:

$$T_{write} = T_{position} + \frac{D}{R_{peak}} \tag{43.1}$$

And thus the effective *rate* of writing ($R_{effective}$), which is just the amount of data written divided by the total time to write it, is:

$$R_{effective} = \frac{D}{T_{write}} = \frac{D}{T_{position} + \frac{D}{R_{peak}}}. \tag{43.2}$$

What we're interested in is getting the effective rate ($R_{effective}$) close to the peak rate. Specifically, we want the effective rate to be some fraction F of the peak rate, where $0 < F < 1$ (a typical F might be 0.9, or 90% of the peak rate). In mathematical form, this means we want $R_{effective} = F \times R_{peak}$.

At this point, we can solve for D:

$$R_{effective} = \frac{D}{T_{position} + \frac{D}{R_{peak}}} = F \times R_{peak} \tag{43.3}$$

$$D = F \times R_{peak} \times (T_{position} + \frac{D}{R_{peak}}) \tag{43.4}$$

$$D = (F \times R_{peak} \times T_{position}) + (F \times R_{peak} \times \frac{D}{R_{peak}}) \tag{43.5}$$

$$D = \frac{F}{1 - F} \times R_{peak} \times T_{position} \tag{43.6}$$

Let's do an example, with a disk with a positioning time of 10 milliseconds and peak transfer rate of 100 MB/s; assume we want an effective bandwidth of 90% of peak ($F = 0.9$). In this case, $D = \frac{0.9}{0.1} \times 100\ MB/s \times 0.01\ seconds = 9\ MB$. Try some different values to see how much we need to buffer in order to approach peak bandwidth. How much is needed to reach 95% of peak? 99%?

43.4 Problem: Finding Inodes

To understand how we find an inode in LFS, let us briefly review how to find an inode in a typical UNIX file system. In a typical file system such as FFS, or even the old UNIX file system, finding inodes is easy, because they are organized in an array and placed on disk at fixed locations.

For example, the old UNIX file system keeps all inodes at a fixed portion of the disk. Thus, given an inode number and the start address, to find a particular inode, you can calculate its exact disk address simply by multiplying the inode number by the size of an inode, and adding that to the start address of the on-disk array; array-based indexing, given an inode number, is fast and straightforward.

Finding an inode given an inode number in FFS is only slightly more complicated, because FFS splits up the inode table into chunks and places a group of inodes within each cylinder group. Thus, one must know how big each chunk of inodes is and the start addresses of each. After that, the calculations are similar and also easy.

In LFS, life is more difficult. Why? Well, we've managed to scatter the inodes all throughout the disk! Worse, we never overwrite in place, and thus the latest version of an inode (i.e., the one we want) keeps moving.

43.5 Solution Through Indirection: The Inode Map

To remedy this, the designers of LFS introduced a **level of indirection** between inode numbers and the inodes through a data structure called the **inode map (imap)**. The imap is a structure that takes an inode number as input and produces the disk address of the most recent version of the

TIP: USE A LEVEL OF INDIRECTION

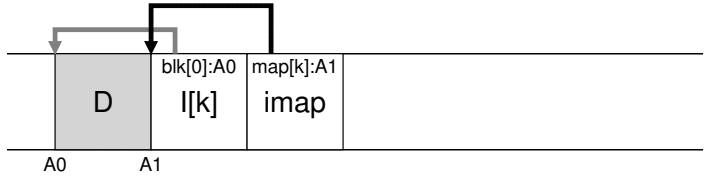
People often say that the solution to all problems in Computer Science is simply a **level of indirection**. This is clearly not true; it is just the solution to *most* problems (yes, this is still too strong of a comment, but you get the point). You certainly can think of every virtualization we have studied, e.g., virtual memory, or the notion of a file, as simply a level of indirection. And certainly the inode map in LFS is a virtualization of inode numbers. Hopefully you can see the great power of indirection in these examples, allowing us to freely move structures around (such as pages in the VM example, or inodes in LFS) without having to change every reference to them. Of course, indirection can have a downside too: **extra overhead.** So next time you have a problem, try solving it with indirection, but make sure to think about the overheads of doing so first. As Wheeler famously said, "All problems in computer science can be solved by another level of indirection, except of course for the problem of too many indirections."

inode. Thus, you can imagine it would often be implemented as a simple *array*, with 4 bytes (a disk pointer) per entry. Any time an inode is written to disk, the imap is updated with its new location.

The imap, unfortunately, needs to be kept persistent (i.e., written to disk); doing so allows LFS to keep track of the locations of inodes across crashes, and thus operate as desired. Thus, a question: where should the imap reside on disk?

It could live on a fixed part of the disk, of course. Unfortunately, as it gets updated frequently, this would then require updates to file structures to be followed by writes to the imap, and hence performance would suffer (i.e., there would be more disk seeks, between each update and the fixed location of the imap).

Instead, LFS places chunks of the inode map right next to where it is writing all of the other new information. Thus, when appending a data block to a file k, LFS actually writes the new data block, its inode, and a piece of the inode map all together onto the disk, as follows:



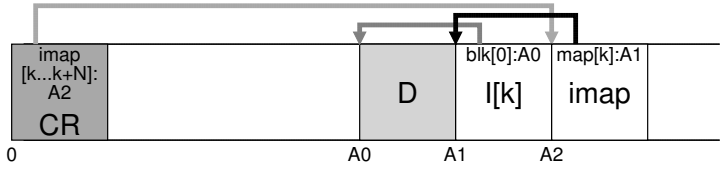
In this picture, the piece of the imap array stored in the block marked *imap* tells LFS that the inode k is at disk address $A1$; this inode, in turn, tells LFS that its data block D is at address $A0$.

43.6 Completing The Solution: The Checkpoint Region

The clever reader (that's you, right?) might have noticed a problem here. How do we find the inode map, now that pieces of it are also now spread across the disk? In the end, there is no magic: the file system must have *some* fixed and known location on disk to begin a file lookup.

LFS has just such a fixed place on disk for this, known as the **checkpoint region (CR)**. The checkpoint region contains pointers to (i.e., addresses of) the latest pieces of the inode map, and thus the inode map pieces can be found by reading the CR first. Note the checkpoint region is only updated periodically (say every 30 seconds or so), and thus performance is not ill-affected. Thus, the overall structure of the on-disk layout contains a checkpoint region (which points to the latest pieces of the inode map); the inode map pieces each contain addresses of the inodes; the inodes point to files (and directories) just like typical UNIX file systems.

Here is an example of the checkpoint region (note it is all the way at the beginning of the disk, at address 0), and a single imap chunk, inode, and data block. A real file system would of course have a much bigger CR (indeed, it would have two, as we'll come to understand later), many imap chunks, and of course many more inodes, data blocks, etc.



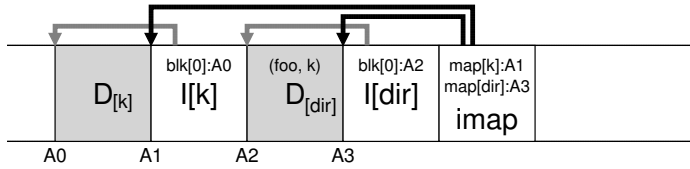
43.7 Reading A File From Disk: A Recap

To make sure you understand how LFS works, let us now walk through what must happen to read a file from disk. Assume we have nothing in memory to begin. The first on-disk data structure we must read is the checkpoint region. The checkpoint region contains pointers (i.e., disk addresses) to the entire inode map, and thus LFS then reads in the entire inode map and caches it in memory. After this point, when given an inode number of a file, LFS simply looks up the inode-number to inode-disk-address mapping in the imap, and reads in the most recent version of the inode. To read a block from the file, at this point, LFS proceeds exactly as a typical UNIX file system, by using direct pointers or indirect pointers or doubly-indirect pointers as need be. In the common case, LFS should perform the same number of I/Os as a typical file system when reading a file from disk; the entire imap is cached and thus the extra work LFS does during a read is to look up the inode's address in the imap.

43.8 What About Directories?

Thus far, we've simplified our discussion a bit by only considering inodes and data blocks. However, to access a file in a file system (such as `/home/remzi/foo`, one of our favorite fake file names), some directories must be accessed too. So how does LFS store directory data?

Fortunately, directory structure is basically identical to classic UNIX file systems, in that a directory is just a collection of (name, inode number) mappings. For example, when creating a file on disk, LFS must both write a new inode, some data, as well as the directory data and its inode that refer to this file. Remember that LFS will do so sequentially on the disk (after buffering the updates for some time). Thus, creating a file `foo` in a directory would lead to the following new structures on disk:



The piece of the inode map contains the information for the location of both the directory file dir as well as the newly-created file f. Thus, when accessing file `foo` (with inode number k), you would first look in the inode map (usually cached in memory) to find the location of the inode of directory dir ($A3$); you then read the directory inode, which gives you the location of the directory data ($A2$); reading this data block gives you the name-to-inode-number mapping of (`foo`, k). You then consult the inode map again to find the location of inode number k ($A1$), and finally read the desired data block at address $A0$.

There is one other serious problem in LFS that the inode map solves, known as the **recursive update problem** [Z+12]. The problem arises in any file system that never updates in place (such as LFS), but rather moves updates to new locations on the disk.

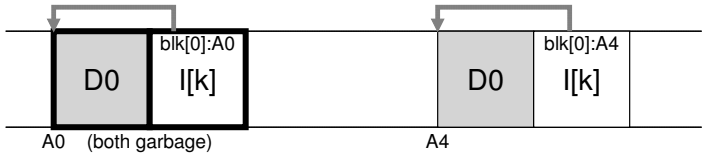
Specifically, whenever an inode is updated, its location on disk changes. If we hadn't been careful, this would have also entailed an update to the directory that points to this file, which then would have mandated a change to the parent of that directory, and so on, all the way up the file system tree.

LFS cleverly avoids this problem with the inode map. Even though the location of an inode may change, the change is never reflected in the directory itself; rather, the imap structure is updated while the directory holds the same name-to-inumber mapping. Thus, through indirection, LFS avoids the recursive update problem.

43.9 A New Problem: Garbage Collection

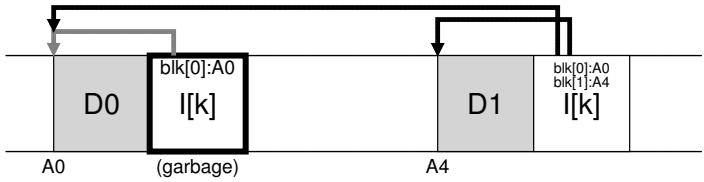
You may have noticed another problem with LFS; it repeatedly writes the latest version of a file (including its inode and data) to new locations on disk. This process, while keeping writes efficient, implies that LFS leaves old versions of file structures scattered throughout the disk. We (rather unceremoniously) call these old versions **garbage**.

For example, let's imagine the case where we have an existing file referred to by inode number k, which points to a single data block $D0$. We now update that block, generating both a new inode and a new data block. The resulting on-disk layout of LFS would look something like this (note we omit the imap and other structures for simplicity; a new chunk of imap would also have to be written to disk to point to the new inode):



In the diagram, you can see that both the inode and data block have two versions on disk, one old (the one on the left) and one current and thus **live** (the one on the right). By the simple act of (logically) updating a data block, a number of new structures must be persisted by LFS, thus leaving old versions of said blocks on the disk.

As another example, imagine we instead append a block to that original file k. In this case, a new version of the inode is generated, but the old data block is still pointed to by the inode. Thus, it is still live and very much part of the current file system:



So what should we do with these older versions of inodes, data blocks, and so forth? One could keep those older versions around and allow users to restore old file versions (for example, when they accidentally overwrite or delete a file, it could be quite handy to do so); such a file system is known as a **versioning file system** because it keeps track of the different versions of a file.

However, LFS instead keeps only the latest live version of a file; thus (in the background), LFS must periodically find these old dead versions of file data, inodes, and other structures, and **clean** them; cleaning should

thus make blocks on disk free again for use in a subsequent writes. Note that the process of cleaning is a form of **garbage collection**, a technique that arises in programming languages that automatically free unused memory for programs.

Earlier we discussed segments as important as they are the mechanism that enables large writes to disk in LFS. As it turns out, they are also quite integral to effective cleaning. Imagine what would happen if the LFS cleaner simply went through and freed single data blocks, inodes, etc., during cleaning. The result: a file system with some number of free **holes** mixed between allocated space on disk. Write performance would drop considerably, as LFS would not be able to find a large contiguous region to write to disk sequentially and with high performance.

Instead, the LFS cleaner works on a segment-by-segment basis, thus clearing up large chunks of space for subsequent writing. The basic cleaning process works as follows. Periodically, the LFS cleaner reads in a number of old (partially-used) segments, determines which blocks are live within these segments, and then write out a new set of segments with just the live blocks within them, freeing up the old ones for writing. Specifically, we expect the cleaner to read in M existing segments, **compact** their contents into N new segments (where $N < M$), and then write the N segments to disk in new locations. The old M segments are then freed and can be used by the file system for subsequent writes.

We are now left with two problems, however. The first is mechanism: how can LFS tell which blocks within a segment are live, and which are dead? The second is policy: how often should the cleaner run, and which segments should it pick to clean?

43.10 Determining Block Liveness

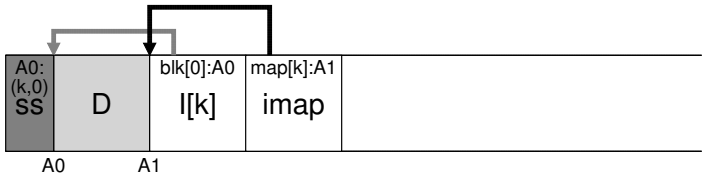
We address the mechanism first. Given a data block D within an on-disk segment S, LFS must be able to determine whether D is live. To do so, LFS adds a little extra information to each segment that describes each block. Specifically, LFS includes, for each data block D, its inode number (which file it belongs to) and its offset (which block of the file this is). This information is recorded in a structure at the head of the segment known as the **segment summary block**.

Given this information, it is straightforward to determine whether a block is live or dead. For a block D located on disk at address A, look in the segment summary block and find its inode number N and offset T. Next, look in the imap to find where N lives and read N from disk (perhaps it is already in memory, which is even better). Finally, using the offset T, look in the inode (or some indirect block) to see where the inode thinks the Tth block of this file is on disk. If it points exactly to disk address A, LFS can conclude that the block D is live. If it points anywhere else, LFS can conclude that D is not in use (i.e., it is dead) and thus know that this version is no longer needed. A pseudocode summary of this

process is shown here:

```
(N, T) = SegmentSummary[A];
inode  = Read(imap[N]);
if (inode[T] == A)
    // block D is alive
else
    // block D is garbage
```

Here is a diagram depicting the mechanism, in which the segment summary block (marked SS) records that the data block at address $A0$ is actually a part of file k at offset 0. By checking the imap for k, you can find the inode, and see that it does indeed point to that location.



There are some shortcuts LFS takes to make the process of determining liveness more efficient. For example, when a file is truncated or deleted, LFS increases its **version number** and records the new version number in the imap. By also recording the version number in the on-disk segment, LFS can short circuit the longer check described above simply by comparing the on-disk version number with a version number in the imap, thus avoiding extra reads.

43.11 A Policy Question: Which Blocks To Clean, And When?

On top of the mechanism described above, LFS must include a set of policies to determine both when to clean and which blocks are worth cleaning. Determining when to clean is easier; either periodically, during idle time, or when you have to because the disk is full.

Determining which blocks to clean is more challenging, and has been the subject of many research papers. In the original LFS paper [RO91], the authors describe an approach which tries to segregate *hot* and *cold* segments. A hot segment is one in which the contents are being frequently over-written; thus, for such a segment, the best policy is to wait a long time before cleaning it, as more and more blocks are getting over-written (in new segments) and thus being freed for use. A cold segment, in contrast, may have a few dead blocks but the rest of its contents are relatively stable. Thus, the authors conclude that one should clean cold segments sooner and hot segments later, and develop a heuristic that does exactly that. However, as with most policies, this policy isn't perfect; later approaches show how to do better [MR+97].

43.12 Crash Recovery And The Log

One final problem: what happens if the system crashes while LFS is writing to disk? As you may recall in the previous chapter about journaling, crashes during updates are tricky for file systems, and thus something LFS must consider as well.

During normal operation, LFS buffers writes in a segment, and then (when the segment is full, or when some amount of time has elapsed), writes the segment to disk. LFS organizes these writes in a **log**, i.e., the checkpoint region points to a head and tail segment, and each segment points to the next segment to be written. LFS also periodically updates the checkpoint region. Crashes could clearly happen during either of these operations (write to a segment, write to the CR). So how does LFS handle crashes during writes to these structures?

Let's cover the second case first. To ensure that the CR update happens atomically, LFS actually keeps two CRs, one at either end of the disk, and writes to them alternately. LFS also implements a careful protocol when updating the CR with the latest pointers to the inode map and other information; specifically, it first writes out a header (with timestamp), then the body of the CR, and then finally one last block (also with a timestamp). If the system crashes during a CR update, LFS can detect this by seeing an inconsistent pair of timestamps. LFS will always choose to use the most recent CR that has consistent timestamps, and thus consistent update of the CR is achieved.

Let's now address the first case. Because LFS writes the CR every 30 seconds or so, the last consistent snapshot of the file system may be quite old. Thus, upon reboot, LFS can easily recover by simply reading in the checkpoint region, the imap pieces it points to, and subsequent files and directories; however, the last many seconds of updates would be lost.

To improve upon this, LFS tries to rebuild many of those segments through a technique known as **roll forward** in the database community. The basic idea is to start with the last checkpoint region, find the end of the log (which is included in the CR), and then use that to read through the next segments and see if there are any valid updates within it. If there are, LFS updates the file system accordingly and thus recovers much of the data and metadata written since the last checkpoint. See Rosenblum's award-winning dissertation for details [R92].

43.13 Summary

LFS introduces a new approach to updating the disk. Instead of overwriting files in places, LFS always writes to an unused portion of the disk, and then later reclaims that old space through cleaning. This approach, which in database systems is called **shadow paging** [L77] and in file-system-speak is sometimes called **copy-on-write**, enables highly efficient writing, as LFS can gather all updates into an in-memory segment and then write them out together sequentially.

TIP: TURN FLAWS INTO VIRTUES

Whenever your system has a fundamental flaw, see if you can turn it around into a feature or something useful. NetApp's WAFL does this with old file contents; by making old versions available, WAFL no longer has to worry about cleaning quite so often (though it does delete old versions, eventually, in the background), and thus provides a cool feature and removes much of the LFS cleaning problem all in one wonderful twist. Are there other examples of this in systems? Undoubtedly, but you'll have to think of them yourself, because this chapter is over with a capital "O". Over. Done. Kaput. We're out. Peace!

The downside to this approach is that it generates garbage; old copies of the data are scattered throughout the disk, and if one wants to reclaim such space for subsequent usage, one must clean old segments periodically. Cleaning became the focus of much controversy in LFS, and concerns over cleaning costs [SS+95] perhaps limited LFS's initial impact on the field. However, some modern commercial file systems, including NetApp's **WAFL** [HLM94], Sun's **ZFS** [B07], and Linux **btrfs** [R+13], and even modern **flash-based SSDs** [AD14], adopt a similar copy-on-write approach to writing to disk, and thus the intellectual legacy of LFS lives on in these modern file systems. In particular, WAFL got around cleaning problems by turning them into a feature; by providing old versions of the file system via **snapshots**, users could access old files whenever they deleted current ones accidentally.

References

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Chapter: Flash-based Solid State Drives
Remzi Arpaci-Dusseau and Andrea Arpaci-Dusseau
A bit gauche to refer you to another chapter in this very book, but who are we to judge?

[B07] "ZFS: The Last Word in File Systems"
Jeff Bonwick and Bill Moore
Copy Available: http://www.ostep.org/Citations/zfs_last.pdf
Slides on ZFS; unfortunately, there is no great ZFS paper (yet). Maybe you will write one, so we can cite it here?

[HLM94] "File System Design for an NFS File Server Appliance"
Dave Hitz, James Lau, Michael Malcolm
USENIX Spring '94
WAFL takes many ideas from LFS and RAID and puts it into a high-speed NFS appliance for the multi-billion dollar storage company NetApp.

[L77] "Physical Integrity in a Large Segmented Database"
R. Lorie
ACM Transactions on Databases, 1977, Volume 2:1, pages 91-104
The original idea of shadow paging is presented here.

[MJLF84] "A Fast File System for UNIX"
Marshall K. McKusick, William N. Joy, Sam J. Leffler, Robert S. Fabry
ACM TOCS, August, 1984, Volume 2, Number 3
The original FFS paper; see the chapter on FFS for more details.

[MR+97] "Improving the Performance of Log-structured File Systems
with Adaptive Methods"
Jeanna Neefe Matthews, Drew Roselli, Adam M. Costello,
Randolph Y. Wang, Thomas E. Anderson
SOSP 1997, pages 238-251, October, Saint Malo, France
A more recent paper detailing better policies for cleaning in LFS.

[M94] "A Better Update Policy"
Jeffrey C. Mogul
USENIX ATC '94, June 1994
In this paper, Mogul finds that read workloads can be harmed by buffering writes for too long and then sending them to the disk in a big burst. Thus, he recommends sending writes more frequently and in smaller batches.

[P98] "Hardware Technology Trends and Database Opportunities"
David A. Patterson
ACM SIGMOD '98 Keynote Address, Presented June 3, 1998, Seattle, Washington
Available: http://www.cs.berkeley.edu/~pattrsn/talks/keynote.html
A great set of slides on technology trends in computer systems. Hopefully, Patterson will create another of these sometime soon.

[R+13] "BTRFS: The Linux B-Tree Filesystem"
Ohad Rodeh, Josef Bacik, Chris Mason
ACM Transactions on Storage, Volume 9 Issue 3, August 2013
Finally, a good paper on BTRFS, a modern take on copy-on-write file systems.

[RO91] "Design and Implementation of the Log-structured File System"
Mendel Rosenblum and John Ousterhout
SOSP '91, Pacific Grove, CA, October 1991
The original SOSP paper about LFS, which has been cited by hundreds of other papers and inspired many real systems.

[R92] "Design and Implementation of the Log-structured File System"
Mendel Rosenblum
http://www.eecs.berkeley.edu/Pubs/TechRpts/1992/CSD-92-696.pdf
The award-winning dissertation about LFS, with many of the details missing from the paper.

[SS+95] "File system logging versus clustering: a performance comparison"
Margo Seltzer, Keith A. Smith, Hari Balakrishnan, Jacqueline Chang, Sara McMains, Venkata Padmanabhan
USENIX 1995 Technical Conference, New Orleans, Louisiana, 1995
A paper that showed the LFS performance sometimes has problems, particularly for workloads with many calls to `fsync()` (such as database workloads). The paper was controversial at the time.

[SO90] "Write-Only Disk Caches"
Jon A. Solworth, Cyril U. Orji
SIGMOD '90, Atlantic City, New Jersey, May 1990
An early study of write buffering and its benefits. However, buffering for too long can be harmful: see Mogul [M94] for details.

[Z+12] "De-indirection for Flash-based SSDs with Nameless Writes"
Yiying Zhang, Leo Prasath Arulraj, Andrea C. Arpaci-Dusseau, Remzi H. Arpaci-Dusseau
FAST '13, San Jose, California, February 2013
Our paper on a new way to build flash-based storage devices. Because FTLs (flash-translation layers) are usually built in a log-structured style, some of the same issues arise in flash-based devices that do in LFS. In this case, it is the recursive update problem, which LFS solves neatly with an imap. A similar structure exists in most SSDs.

Homework

This section introduces `lfs.py`, a simple LFS simulator you can use to understand better how an LFS-based file system works. Read the README for details on how to run the simulator.

Questions

1. Run `./lfs.py -n 3`, perhaps varying the seed (`-s`). Can you figure out which commands were run to generate the final file system contents? Can you tell which order those commands were issued? Finally, can you determine the liveness of each block in the final file system state? Use `-o` to show which commands were run, and `-c` to show the liveness of the final file system state. How much harder does the task become for you as you increase the number of commands issued (i.e., change `-n 3` to `-n 5`)?
2. If you find the above painful, you can help yourself a little bit by showing the set of updates caused by each specific command. To do so, run `./lfs.py -n 3 -i`. Now see if it is easier to understand what each command must have been. Change the random seed to get different commands to interpret (e.g., `-s 1`, `-s 2`, `-s 3`, etc.).
3. To further test your ability to figure out what updates are made to disk by each command, run the following: `./lfs.py -o -F -s 100` (and perhaps a few other random seeds). This just shows a set of commands and does NOT show you the final state of the file system. Can you reason about what the final state of the file system must be?
4. Now see if you can determine which files and directories are live after a number of file and directory operations. Run `tt ./lfs.py -n 20 -s 1` and then examine the final file system state. Can you figure out which pathnames are valid? Run `tt ./lfs.py -n 20 -s 1 -c -v` to see the results. Run with `-o` to see if your answers match up given the series of random commands. Use different random seeds to get more problems.
5. Now let's issue some specific commands. First, let's create a file and write to it repeatedly. To do so, use the `-L` flag, which lets you specify specific commands to execute. In this case, let's create the file "/foo" and write to it 4 times:
   `./lfs.py -L c,/foo:w,/foo,0,1:w,/foo,1,1:w,/foo,2,1:w,/foo,3,1 -o`. See if you can determine the liveness of the final file system state; use `-c` to check your answers.
6. Now, let's do the same thing, but with a single write operation instead of four. Run `./lfs.py -o -L c,/foo:w,/foo,0,4` to create file "/foo" and write 4 blocks with a single write operation. Compute the liveness again, and check if you are right with `-c`. What is the main difference between writing a file all at once (as we do here) versus doing it one block at a time (as above)? What

does this tell you about the importance of buffering updates in main memory as the real LFS does?

7. Let's do another specific example. First, run the following command: `./lfs.py -L c,/foo:w,/foo,0,1`. What does this set of commands do? Now, run `./lfs.py -L c,/foo:w,/foo,7,1`. What does this set of commands do? How are the two different? What can you tell about the `size` field in the inode from these two sets of commands?
8. Now let's look explicitly at file creation versus directory creation. Run `./lfs.py -L c,/foo` and `./lfs.py -L d,/foo` to create a file and then a directory. What is similar about these runs, and what is different?
9. The LFS simulator supports hard links as well. Run the following command to study how they work: `./lfs.py -L c,/foo:l,/foo,/bar:l,/foo,/goo -o -i`. What blocks are written out when a hard link is created? How is this similar to just creating a new file, and how is it different? How does the reference count field change as links are created?
10. LFS makes many different policy decisions. We do not explore many of them here – perhaps something left for the future – but here is a simple one we do explore: the choice of inode number. First, run `./lfs.py -p c100 -n 10 -o -a s` to show the usual behavior with the "sequential" allocation policy, which tries to use free inode numbers nearest to zero. Then, change to a "random" policy by running `./lfs.py -p c100 -n 10 -o -a r` (the `-p c100` flag ensures 100 percent of the random operations are file creations). What on-disk differences does a random policy versus a sequential policy result in? What does this say about the importance of choosing inode numbers in a real LFS?
11. One last thing we've been assuming is that the LFS simulator always updates the checkpoint region after each update. In the real LFS, that isn't the case: it is updated periodically to avoid long seeks. Run `./lfs.py -N -i -o -s 1000` to see some operations and the intermediate and final states of the file system when the checkpoint region isn't forced to disk. What would happen if the checkpoint region is never updated? What if it is updated periodically? Could you figure out how to recover the file system to the latest state by rolling forward in the log?

44

Data Integrity and Protection

Beyond the basic advances found in the file systems we have studied thus far, a number of features are worth studying. In this chapter, we focus on reliability once again (having previously studied storage system reliability in the RAID chapter). Specifically, how should a file system or storage system ensure that data is safe, given the unreliable nature of modern storage devices?

This general area is referred to as **data integrity** or **data protection**. Thus, we will now investigate techniques used to ensure that the data you put into your storage system is the same when the storage system returns it to you.

CRUX: HOW TO ENSURE DATA INTEGRITY

How should systems ensure that the data written to storage is protected? What techniques are required? How can such techniques be made efficient, with both low space and time overheads?

44.1 Disk Failure Modes

As you learned in the chapter about RAID, disks are not perfect, and can fail (on occasion). In early RAID systems, the model of failure was quite simple: either the entire disk is working, or it fails completely, and the detection of such a failure is straightforward. This **fail-stop** model of disk failure makes building RAID relatively simple [S90].

What you didn't learn is about all of the other types of failure modes modern disks exhibit. Specifically, as Bairavasundaram et al. studied in great detail [B+07, B+08], modern disks will occasionally seem to be mostly working but have trouble successfully accessing one or more blocks. Specifically, two types of single-block failures are common and worthy of consideration: **latent-sector errors** (**LSEs**) and **block corruption**. We'll now discuss each in more detail.

	Cheap	Costly
LSEs	9.40%	1.40%
Corruption	0.50%	0.05%

Figure 44.1: **Frequency Of LSEs And Block Corruption**

LSEs arise when a disk sector (or group of sectors) has been damaged in some way. For example, if the disk head touches the surface for some reason (a **head crash**, something which shouldn't happen during normal operation), it may damage the surface, making the bits unreadable. Cosmic rays can also flip bits, leading to incorrect contents. Fortunately, in-disk **error correcting codes** (**ECC**) are used by the drive to determine whether the on-disk bits in a block are good, and in some cases, to fix them; if they are not good, and the drive does not have enough information to fix the error, the disk will return an error when a request is issued to read them.

There are also cases where a disk block becomes **corrupt** in a way not detectable by the disk itself. For example, buggy disk firmware may write a block to the wrong location; in such a case, the disk ECC indicates the block contents are fine, but from the client's perspective the wrong block is returned when subsequently accessed. Similarly, a block may get corrupted when it is transferred from the host to the disk across a faulty bus; the resulting corrupt data is stored by the disk, but it is not what the client desires. These types of faults are particularly insidious because they are **silent faults**; the disk gives no indication of the problem when returning the faulty data.

Prabhakaran et al. describes this more modern view of disk failure as the **fail-partial** disk failure model [P+05]. In this view, disks can still fail in their entirety (as was the case in the traditional fail-stop model); however, disks can also seemingly be working and have one or more blocks become inaccessible (i.e., LSEs) or hold the wrong contents (i.e., corruption). Thus, when accessing a seemingly-working disk, once in a while it may either return an error when trying to read or write a given block (a non-silent partial fault), and once in a while it may simply return the wrong data (a silent partial fault).

Both of these types of faults are somewhat rare, but just how rare? Figure 44.1 summarizes some of the findings from the two Bairavasundaram studies [B+07,B+08].

The figure shows the percent of drives that exhibited at least one LSE or block corruption over the course of the study (about 3 years, over 1.5 million disk drives). The figure further sub-divides the results into "cheap" drives (usually SATA drives) and "costly" drives (usually SCSI or FibreChannel). As you can see, while buying better drives reduces the frequency of both types of problem (by about an order of magnitude), they still happen often enough that you need to think carefully about how to handle them in your storage system.

Some additional findings about LSEs are:

- Costly drives with more than one LSE are as likely to develop additional errors as cheaper drives
- For most drives, annual error rate increases in year two
- The number of LSEs increase with disk size
- Most disks with LSEs have less than 50
- Disks with LSEs are more likely to develop additional LSEs
- There exists a significant amount of spatial and temporal locality
- Disk scrubbing is useful (most LSEs were found this way)

Some findings about corruption:

- Chance of corruption varies greatly across different drive models within the same drive class
- Age effects are different across models
- Workload and disk size have little impact on corruption
- Most disks with corruption only have a few corruptions
- Corruption is not independent within a disk or across disks in RAID
- There exists spatial locality, and some temporal locality
- There is a weak correlation with LSEs

To learn more about these failures, you should likely read the original papers [B+07,B+08]. But hopefully the main point should be clear: if you really wish to build a reliable storage system, you must include machinery to detect and recover from both LSEs and block corruption.

44.2 Handling Latent Sector Errors

Given these two new modes of partial disk failure, we should now try to see what we can do about them. Let's first tackle the easier of the two, namely latent sector errors.

CRUX: HOW TO HANDLE LATENT SECTOR ERRORS
How should a storage system handle latent sector errors? How much extra machinery is needed to handle this form of partial failure?

As it turns out, latent sector errors are rather straightforward to handle, as they are (by definition) easily detected. When a storage system tries to access a block, and the disk returns an error, the storage system should simply use whatever redundancy mechanism it has to return the correct data. In a mirrored RAID, for example, the system should access the alternate copy; in a RAID-4 or RAID-5 system based on parity, the system should reconstruct the block from the other blocks in the parity group. Thus, easily detected problems such as LSEs are readily recovered through standard redundancy mechanisms.

The growing prevalence of LSEs has influenced RAID designs over the years. One particularly interesting problem arises in RAID-4/5 systems when both full-disk faults and LSEs occur in tandem. Specifically, when an entire disk fails, the RAID tries to **reconstruct** the disk (say, onto a hot spare) by reading through all of the other disks in the parity group and recomputing the missing values. If, during reconstruction, an LSE is encountered on any one of the other disks, we have a problem: the reconstruction cannot successfully complete.

To combat this issue, some systems add an extra degree of redundancy. For example, NetApp's **RAID-DP** has the equivalent of two parity disks instead of one [C+04]. When an LSE is discovered during reconstruction, the extra parity helps to reconstruct the missing block. As always, there is a cost, in that maintaining two parity blocks for each stripe is more costly; however, the log-structured nature of the NetApp **WAFL** file system mitigates that cost in many cases [HLM94]. The remaining cost is space, in the form of an extra disk for the second parity block.

44.3 Detecting Corruption: The Checksum

Let's now tackle the more challenging problem, that of silent failures via data corruption. How can we prevent users from getting bad data when corruption arises, and thus leads to disks returning bad data?

CRUX: HOW TO PRESERVE DATA INTEGRITY DESPITE CORRUPTION
Given the silent nature of such failures, what can a storage system do to detect when corruption arises? What techniques are needed? How can one implement them efficiently?

Unlike latent sector errors, *detection* of corruption is a key problem. How can a client tell that a block has gone bad? Once it is known that a particular block is bad, *recovery* is the same as before: you need to have some other copy of the block around (and hopefully, one that is not corrupt!). Thus, we focus here on detection techniques.

The primary mechanism used by modern storage systems to preserve data integrity is called the **checksum**. A checksum is simply the result of a function that takes a chunk of data (say a 4KB block) as input and computes a function over said data, producing a small summary of the contents of the data (say 4 or 8 bytes). This summary is referred to as the checksum. The goal of such a computation is to enable a system to detect if data has somehow been corrupted or altered by storing the checksum with the data and then confirming upon later access that the data's current checksum matches the original storage value.

TIP: THERE'S NO FREE LUNCH

There's No Such Thing As A Free Lunch, or TNSTAAFL for short, is an old American idiom that implies that when you are seemingly getting something for free, in actuality you are likely paying some cost for it. It comes from the old days when diners would advertise a free lunch for customers, hoping to draw them in; only when you went in, did you realize that to acquire the "free" lunch, you had to purchase one or more alcoholic beverages. Of course, this may not actually be a problem, particularly if you are an aspiring alcoholic (or typical undergraduate student).

Common Checksum Functions

A number of different functions are used to compute checksums, and vary in strength (i.e., how good they are at protecting data integrity) and speed (i.e., how quickly can they be computed). A trade-off that is common in systems arises here: usually, the more protection you get, the costlier it is. There is no such thing as a free lunch.

One simple checksum function that some use is based on exclusive or (XOR). With XOR-based checksums, the checksum is computed by XOR'ing each chunk of the data block being checksummed, thus producing a single value that represents the XOR of the entire block.

To make this more concrete, imagine we are computing a 4-byte checksum over a block of 16 bytes (this block is of course too small to really be a disk sector or block, but it will serve for the example). The 16 data bytes, in hex, look like this:

```
365e c4cd ba14 8a92 ecef 2c3a 40be f666
```

If we view them in binary, we get the following:

```
0011 0110 0101 1110    1100 0100 1100 1101
1011 1010 0001 0100    1000 1010 1001 0010
1110 1100 1110 1111    0010 1100 0011 1010
0100 0000 1011 1110    1111 0110 0110 0110
```

Because we've lined up the data in groups of 4 bytes per row, it is easy to see what the resulting checksum will be: perform an XOR over each column to get the final checksum value:

```
0010 0000 0001 1011    1001 0100 0000 0011
```

The result, in hex, is 0x201b9403.

XOR is a reasonable checksum but has its limitations. If, for example, two bits in the same position within each checksummed unit change, the checksum will not detect the corruption. For this reason, people have investigated other checksum functions.

Another basic checksum function is addition. This approach has the advantage of being fast; computing it just requires performing 2's-complement addition over each chunk of the data, ignoring overflow. It can detect many changes in data, but is not good if the data, for example, is shifted.

A slightly more complex algorithm is known as the **Fletcher checksum**, named (as you might guess) for the inventor, John G. Fletcher [F82]. It is quite simple to compute and involves the computation of two check bytes, $s1$ and $s2$. Specifically, assume a block D consists of bytes $d1$ ... dn; $s1$ is defined as follows: $s1 = (s1 + d_i) \ mod \ 255$ (computed over all d_i); $s2$ in turn is: $s2 = (s2 + s1) \ mod \ 255$ (again over all d_i) [F04]. The Fletcher checksum is almost as strong as the CRC (see below), detecting all single-bit and double-bit errors many burst errors [F04].

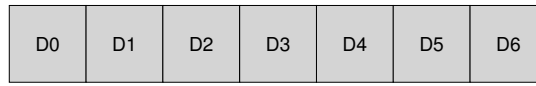
One final commonly-used checksum is known as a **cyclic redundancy check** (**CRC**). Assume you wish to compute the checksum over a data block D. All you do is treat D as if it is a large binary number (it is just a string of bits after all) and divide it by an agreed upon value (k). The remainder of this division is the value of the CRC. As it turns out, one can implement this binary modulo operation rather efficiently, and hence the popularity of the CRC in networking as well. See elsewhere for more details [M13].

Whatever the method used, it should be obvious that there is no perfect checksum: it is possible two data blocks with non-identical contents will have identical checksums, something referred to as a **collision**. This fact should be intuitive: after all, computing a checksum is taking something large (e.g., 4KB) and producing a summary that is much smaller (e.g., 4 or 8 bytes). In choosing a good checksum function, we are thus trying to find one that minimizes the chance of collisions while remaining easy to compute.

Checksum Layout

Now that you understand a bit about how to compute a checksum, let's next analyze how to use checksums in a storage system. The first question we must address is the layout of the checksum, i.e., how should checksums be stored on disk?

The most basic approach simply stores a checksum with each disk sector (or block). Given a data block D, let us call the checksum over that data $C(D)$. Thus, without checksums, the disk layout looks like this:



With checksums, the layout adds a single checksum for every block:

Because checksums are usually small (e.g., 8 bytes), and disks only can write in sector-sized chunks (512 bytes) or multiples thereof, one problem that arises is how to achieve the above layout. One solution employed by drive manufacturers is to format the drive with 520-byte sectors; an extra 8 bytes per sector can be used to store the checksum.

In disks that don't have such functionality, the file system must figure out a way to store the checksums packed into 512-byte blocks. One such possibility is as follows:

In this scheme, the n checksums are stored together in a sector, followed by n data blocks, followed by another checksum sector for the next n blocks, and so forth. This scheme has the benefit of working on all disks, but can be less efficient; if the file system, for example, wants to overwrite block $D1$, it has to read in the checksum sector containing $C(D1)$, update $C(D1)$ in it, and then write out the checksum sector as well as the new data block $D1$ (thus, one read and two writes). The earlier approach (of one checksum per sector) just performs a single write.

44.4 Using Checksums

With a checksum layout decided upon, we can now proceed to actually understand how to *use* the checksums. When reading a block D, the client (i.e., file system or storage controller) also reads its checksum from disk $C_s(D)$, which we call the **stored checksum** (hence the subscript C_s). The client then *computes* the checksum over the retrieved block D, which we call the **computed checksum** $C_c(D)$. At this point, the client compares the stored and computed checksums; if they are equal (i.e., $C_s(D)$ == $C_c(D)$, the data has likely not been corrupted, and thus can be safely returned to the user. If they do *not* match (i.e., $C_s(D)$!= $C_c(D)$), this implies the data has changed since the time it was stored (since the stored checksum reflects the value of the data at that time). In this case, we have a corruption, which our checksum has helped us to detect.

Given a corruption, the natural question is what should we do about it? If the storage system has a redundant copy, the answer is easy: try to use it instead. If the storage system has no such copy, the likely answer is to return an error. In either case, realize that corruption detection is not a magic bullet; if there is no other way to get the non-corrupted data, you are simply out of luck.

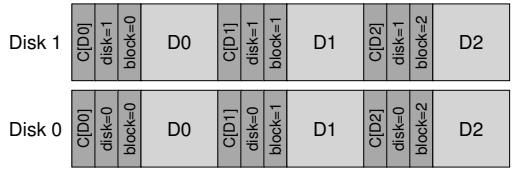
44.5 A New Problem: Misdirected Writes

The basic scheme described above works well in the general case of corrupted blocks. However, modern disks have a couple of unusual failure modes that require different solutions.

The first failure mode of interest is called a **misdirected write**. This arises in disk and RAID controllers which write the data to disk correctly, except in the *wrong* location. In a single-disk system, this means that the disk wrote block D_x not to address x (as desired) but rather to address y (thus "corrupting" D_y); in addition, within a multi-disk system, the controller may also write $D_{i,x}$ not to address x of disk i but rather to some other disk j. Thus our question:

CRUX: HOW TO HANDLE MISDIRECTED WRITES
How should a storage system or disk controller detect misdirected writes? What additional features are required from the checksum?

The answer, not surprisingly, is simple: add a little more information to each checksum. In this case, adding a **physical identifier** (**physical ID**) is quite helpful. For example, if the stored information now contains the checksum $C(D)$ as well as the disk and sector number of the block, it is easy for the client to determine whether the correct information resides within the block. Specifically, if the client is reading block 4 on disk 10 ($D_{10,4}$), the stored information should include that disk number and sector offset, as shown below. If the information does not match, a misdirected write has taken place, and a corruption is now detected. Here is an example of what this added information would look like on a two-disk system. Note that this figure, like the others before it, is not to scale, as the checksums are usually small (e.g., 8 bytes) whereas the blocks are much larger (e.g., 4 KB or bigger):



You can see from the on-disk format that there is now a fair amount of redundancy on disk: for each block, the disk number is repeated within each block, and the offset of the block in question is also kept next to the block itself. The presence of redundant information should be no surprise, though; redundancy is the key to error detection (in this case) and recovery (in others). A little extra information, while not strictly needed with perfect disks, can go a long ways in helping detect problematic situations should they arise.

44.6 One Last Problem: Lost Writes

Unfortunately, misdirected writes are not the last problem we will address. Specifically, some modern storage devices also have an issue known as a **lost write**, which occurs when the device informs the upper layer that a write has completed but in fact it never is persisted; thus, what remains is left is the old contents of the block rather than the updated new contents.

The obvious question here is: do any of our checksumming strategies from above (e.g., basic checksums, or physical identity) help to detect lost writes? Unfortunately, the answer is no: the old block likely has a matching checksum, and the physical ID used above (disk number and block offset) will also be correct. Thus our final problem:

CRUX: HOW TO HANDLE LOST WRITES
How should a storage system or disk controller detect lost writes? What additional features are required from the checksum?

There are a number of possible solutions that can help [K+08]. One classic approach [BS04] is to perform a **write verify** or **read-after-write**; by immediately reading back the data after a write, a system can ensure that the data indeed reached the disk surface. This approach, however, is quite slow, doubling the number of I/Os needed to complete a write.

Some systems add a checksum elsewhere in the system to detect lost writes. For example, Sun's **Zettabyte File System** (**ZFS**) includes a checksum in each file system inode and indirect block for every block included within a file. Thus, even if the write to a data block itself is lost, the checksum within the inode will not match the old data. Only if the writes to both the inode and the data are lost simultaneously will such a scheme fail, an unlikely (but unfortunately, possible!) situation.

44.7 Scrubbing

Given all of this discussion, you might be wondering: when do these checksums actually get checked? Of course, some amount of checking occurs when data is accessed by applications, but most data is rarely accessed, and thus would remain unchecked. Unchecked data is problematic for a reliable storage system, as bit rot could eventually affect all copies of a particular piece of data.

To remedy this problem, many systems utilize **disk scrubbing** of various forms [K+08]. By periodically reading through *every* block of the system, and checking whether checksums are still valid, the disk system can reduce the chances that all copies of a certain data item become corrupted. Typical systems schedule scans on a nightly or weekly basis.

44.8 Overheads Of Checksumming

Before closing, we now discuss some of the overheads of using checksums for data protection. There are two distinct kinds of overheads, as is common in computer systems: space and time.

Space overheads come in two forms. The first is on the disk (or other storage medium) itself; each stored checksum takes up room on the disk, which can no longer be used for user data. A typical ratio might be an 8-byte checksum per 4 KB data block, for a 0.19% on-disk space overhead.

The second type of space overhead comes in the memory of the system. When accessing data, there must now be room in memory for the checksums as well as the data itself. However, if the system simply checks the checksum and then discards it once done, this overhead is short-lived and not much of a concern. Only if checksums are kept in memory (for an added level of protection against memory corruption [Z+13]) will this small overhead be observable.

While space overheads are small, the time overheads induced by checksumming can be quite noticeable. Minimally, the CPU must compute the checksum over each block, both when the data is stored (to determine the value of the stored checksum) as well as when it is accessed (to compute the checksum again and compare it against the stored checksum). One approach to reducing CPU overheads, employed by many systems that use checksums (including network stacks), is to combine data copying and checksumming into one streamlined activity; because the copy is needed anyhow (e.g., to copy the data from the kernel page cache into a user buffer), combined copying/checksumming can be quite effective.

Beyond CPU overheads, some checksumming schemes can induce extra I/O overheads, particularly when checksums are stored distinctly from the data (thus requiring extra I/Os to access them), and for any extra I/O needed for background scrubbing. The former can be reduced by design; the latter can be tuned and thus its impact limited, perhaps by controlling when such scrubbing activity takes place. The middle of the night, when most (not all!) productive workers have gone to bed, may be a good time to perform such scrubbing activity and increase the robustness of the storage system.

44.9 Summary

We have discussed data protection in modern storage systems, focusing on checksum implementation and usage. Different checksums protect against different types of faults; as storage devices evolve, new failure modes will undoubtedly arise. Perhaps such change will force the research community and industry to revisit some of these basic approaches, or invent entirely new approaches altogether. Time will tell. Or it won't. Time is funny that way.

References

[B+07] "An Analysis of Latent Sector Errors in Disk Drives"
Lakshmi N. Bairavasundaram, Garth R. Goodson, Shankar Pasupathy, Jiri Schindler
SIGMETRICS '07, San Diego, California, June 2007
The first paper to study latent sector errors in detail. As described in the next citation [B+08], a collaboration between Wisconsin and NetApp. The paper also won the Kenneth C. Sevcik Outstanding Student Paper award; Sevcik was a terrific researcher and wonderful guy who passed away too soon. To show the authors it was possible to move from the U.S. to Canada and love it, he once sang us the Canadian national anthem, standing up in the middle of a restaurant to do so.

[B+08] "An Analysis of Data Corruption in the Storage Stack"
Lakshmi N. Bairavasundaram, Garth R. Goodson, Bianca Schroeder,
Andrea C. Arpaci-Dusseau, Remzi H. Arpaci-Dusseau
FAST '08, San Jose, CA, February 2008
The first paper to truly study disk corruption in great detail, focusing on how often such corruption occurs over three years for over 1.5 million drives. Lakshmi did this work while a graduate student at Wisconsin under our supervision, but also in collaboration with his colleagues at NetApp where he was an intern for multiple summers. A great example of how working with industry can make for much more interesting and relevant research.

[BS04] "Commercial Fault Tolerance: A Tale of Two Systems"
Wendy Bartlett, Lisa Spainhower
IEEE Transactions on Dependable and Secure Computing, Vol. 1, No. 1, January 2004
This classic in building fault tolerant systems is an excellent overview of the state of the art from both IBM and Tandem. Another must read for those interested in the area.

[C+04] "Row-Diagonal Parity for Double Disk Failure Correction"
P. Corbett, B. English, A. Goel, T. Grcanac, S. Kleiman, J. Leong, S. Sankar
FAST '04, San Jose, CA, February 2004
An early paper on how extra redundancy helps to solve the combined full-disk-failure/partial-disk-failure problem. Also a nice example of how to mix more theoretical work with practical.

[F04] "Checksums and Error Control"
Peter M. Fenwick
Copy Available: http://www.ostep.org/Citations/checksums-03.pdf
A great simple tutorial on checksums, available to you for the amazing cost of free.

[F82] "An Arithmetic Checksum for Serial Transmissions"
John G. Fletcher
IEEE Transactions on Communication, Vol. 30, No. 1, January 1982
*Fletcher's original work on his eponymous checksum. Of course, he didn't call it the Fletcher checksum, rather he just didn't call it anything, and thus it became natural to name it after the inventor. So don't blame old Fletch for this seeming act of braggadocio. This anecdote might remind you of Rubik and his cube; Rubik never called it "**Rubik's cube**"; rather, he just called it "my cube."*

[HLM94] "File System Design for an NFS File Server Appliance"
Dave Hitz, James Lau, Michael Malcolm
USENIX Spring '94
The pioneering paper that describes the ideas and product at the heart of NetApp's core. Based on this system, NetApp has grown into a multi-billion dollar storage company. If you're interested in learning more about its founding, read Hitz's autobiography "How to Castrate a Bull: Unexpected Lessons on Risk, Growth, and Success in Business" (which is the actual title, no joking). And you thought you could avoid bull castration by going into Computer Science.

[K+08] "Parity Lost and Parity Regained"
Andrew Krioukov, Lakshmi N. Bairavasundaram, Garth R. Goodson, Kiran Srinivasan, Randy Thelen, Andrea C. Arpaci-Dusseau, Remzi H. Arpaci-Dusseau
FAST '08, San Jose, CA, February 2008
This work of ours, joint with colleagues at NetApp, explores how different checksum schemes work (or don't work) in protecting data. We reveal a number of interesting flaws in current protection strategies, some of which have led to fixes in commercial products.

[M13] "Cyclic Redundancy Checks"
Author Unknown
Available: http://www.mathpages.com/home/kmath458.htm
Not sure who wrote this, but a super clear and concise description of CRCs is available here. The internet is full of information, as it turns out.

[P+05] "IRON File Systems"
Vijayan Prabhakaran, Lakshmi N. Bairavasundaram, Nitin Agrawal, Haryadi S. Gunawi, Andrea C. Arpaci-Dusseau, Remzi H. Arpaci-Dusseau
SOSP '05, Brighton, England, October 2005
Our paper on how disks have partial failure modes, which includes a detailed study of how file systems such as Linux ext3 and Windows NTFS react to such failures. As it turns out, rather poorly! We found numerous bugs, design flaws, and other oddities in this work. Some of this has fed back into the Linux community, thus helping to yield a new more robust group of file systems to store your data.

[RO91] "Design and Implementation of the Log-structured File System"
Mendel Rosenblum and John Ousterhout
SOSP '91, Pacific Grove, CA, October 1991
Another reference to this ground-breaking paper on how to improve write performance in file systems.

[S90] "Implementing Fault-Tolerant Services Using The State Machine Approach: A Tutorial"
Fred B. Schneider
ACM Surveys, Vol. 22, No. 4, December 1990
This classic paper talks generally about how to build fault tolerant services, and includes many basic definitions of terms. A must read for those building distributed systems.

[Z+13] "Zettabyte Reliability with Flexible End-to-end Data Integrity"
Yupu Zhang, Daniel S. Myers, Andrea C. Arpaci-Dusseau, Remzi H. Arpaci-Dusseau
MSST '13, Long Beach, California, May 2013
Our own work on adding data protection to the page cache of a system, which protects against memory corruption as well as on-disk corruption.

Homework

In this homework, you'll use `checksum.py` to investigate various aspects of checksums.

Questions

1. First just run `checksum.py` with no arguments. Compute the additive, XOR-based, and Fletcher checksums. Use `-c` to check your answers.
2. Now do the same, but vary the seed (`-s`) to different values.
3. Sometimes the additive and XOR-based checksums produce the same checksum (e.g., if the data value is all zeroes). Can you pass in a 4-byte data value (using the `-D` flag, e.g., `-D a,b,c,d`) that does not contain only zeroes and leads the additive and XOR-based checksum having the same value? In general, when does this occur? Check that you are correct with the `-c` flag.
4. Now pass in a 4-byte value that you know will produce a different checksum values for additive and XOR. In general, when does this occur?
5. Use the simulator to compute checksums twice (once each for a different set of numbers). The two number strings should be different (e.g., `-D a1,b1,c1,d1` the first time and `-D a2,b2,c2,d2` the second) but should produce the same additive checksum. In general, when will the additive checksum be the same, even though the data values are different? Check your specific answer with the `-c` flag.
6. Now do the same for the XOR checksum.
7. Now let's look at a specific set of data values. The first is: `-D 1,2,3,4`. What will the different checksums (additive, XOR, Fletcher) be for this data? Now compare it to computing these checksums over `-D 4,3,2,1`. What do you notice about these three checksums? How does Fletcher compare to the other two? How is Fletcher generally "better" than something like the simple additive checksum?
8. No checksum is perfect. Given a particular input of your choosing, can you find other data values that lead to the same Fletcher checksum? When, in general, does this occur? Start with a simple data string (e.g., `-D 0,1,2,3`) and see if you can replace one of those numbers but end up with the same Fletcher checksum. As always, use `-c` to check your answers.

Homework (Code)

In this part of the homework, you'll write some of your own code to implement various checksums.

Questions

1. Write a short C program (called `check-xor.c`) that computes an XOR-based checksum over an input file, and prints the checksum as output. Use a 8-bit unsigned char to store the (one byte) checksum. Make some test files to see if it works as expected.
2. Now write a short C program (called `check-fletcher.c`) that computes the Fletcher checksum over an input file. Once again, test your program to see if it works.
3. Now compare the performance of both: is one faster than the other? How does performance change as the size of the input file changes? Use internal calls to `gettimeofday` to time the programs. Which should you use if you care about performance? About checking ability?
4. Read about the 16-bit CRC and then implement it. Test it on a number of different inputs to ensure that it works. How is its performance as compared to the simple XOR and Fletcher? How about its checking ability?
5. Now build a tool (`create-csum.c`) that computes a single-byte checksum for every 4KB block of a file, and records the results in an output file (specified on the command line). Build a related tool (`check-csum.c`) that reads a file, computes the checksums over each block, and compares the results to the stored checksums stored in another file. If there is a problem, the program should print that the file has been corrupted. Test the program by manually corrupting the file.

45

Summary Dialogue on Persistence

Student: *Wow, file systems seem interesting(!), and yet complicated.*

Professor: *That's why me and my spouse do our research in this space.*

Student: *Hold on. Are you one of the professors who wrote this book? I thought we were both just fake constructs, used to summarize some main points, and perhaps add a little levity in the study of operating systems.*

Professor: *Uh... er... maybe. And none of your business! And who did you think was writing these things? (sighs) Anyhow, let's get on with it: what did you learn?*

Student: *Well, I think I got one of the main points, which is that it is much harder to manage data for a long time (persistently) than it is to manage data that isn't persistent (like the stuff in memory). After all, if your machines crashes, memory contents disappear! But the stuff in the file system needs to live forever.*

Professor: *Well, as my friend Kevin Hultquist used to say, "Forever is a long time"; while he was talking about plastic golf tees, it's especially true for the garbage that is found in most file systems.*

Student: *Well, you know what I mean! For a long time at least. And even simple things, such as updating a persistent storage device, are complicated, because you have to care what happens if you crash. Recovery, something I had never even thought of when we were virtualizing memory, is now a big deal!*

Professor: *Too true. Updates to persistent storage have always been, and remain, a fun and challenging problem.*

Student: *I also learned about cool things like disk scheduling, and about data protection techniques like RAID and even checksums. That stuff is cool.*

Professor: *I like those topics too. Though, if you really get into it, they can get a little mathematical. Check out some the latest on erasure codes if you want your brain to hurt.*

Student: *I'll get right on that.*

Professor: *(frowns) I think you're being sarcastic. Well, what else did you like?*

Student: *And I also liked all the thought that has gone into building technology-aware systems, like FFS and LFS. Neat stuff! Being disk aware seems cool. But will it matter anymore, with Flash and all the newest, latest technologies?*

Professor: *Good question! And a reminder to get working on that Flash chapter... (scribbles note down to self) ... But yes, even with Flash, all of this stuff is still relevant, amazingly. For example, Flash Translation Layers (FTLs) use log-structuring internally, to improve performance and reliability of Flash-based SSDs. And thinking about locality is always useful. So while the technology may be changing, many of the ideas we have studied will continue to be useful, for a while at least.*

Student: *That's good. I just spent all this time learning it, and I didn't want it to all be for no reason!*

Professor: *Professors wouldn't do that to you, would they?*

46

A Dialogue on Distribution

Professor: *And thus we reach our final little piece in the world of operating systems: distributed systems. Since we can't cover much here, we'll sneak in a little intro here in the section on persistence, and focus mostly on distributed file systems. Hope that is OK!*

Student: *Sounds OK. But what is a distributed system exactly, oh glorious and all-knowing professor?*

Professor: *Well, I bet you know how this is going to go...*

Student: *There's a peach?*

Professor: *Exactly! But this time, it's far away from you, and may take some time to get the peach. And there are a lot of them! Even worse, sometimes a peach becomes rotten. But you want to make sure that when anybody bites into a peach, they will get a mouthful of deliciousness.*

Student: *This peach analogy is working less and less for me.*

Professor: *Come on! It's the last one, just go with it.*

Student: *Fine.*

Professor: *So anyhow, forget about the peaches. Building distributed systems is hard, because things fail all the time. Messages get lost, machines go down, disks corrupt data. It's like the whole world is working against you!*

Student: *But I use distributed systems all the time, right?*

Professor: *Yes! You do. And... ?*

Student: *Well, it seems like they mostly work. After all, when I send a search request to Google, it usually comes back in a snap, with some great results! Same thing when I use Facebook, Amazon, and so forth.*

Professor: *Yes, it is amazing. And that's despite all of those failures taking place! Those companies build a huge amount of machinery into their systems so as to ensure that even though some machines have failed, the entire system stays up and running. They use a lot of techniques to do this: replication, retry, and various other tricks people have developed over time to detect and recover from failures.*

Student: *Sounds interesting. Time to learn something for real?*

Professor: *It does seem so. Let's get to work! But first things first ...*
(bites into peach he has been holding, which unfortunately is rotten)

47

Distributed Systems

Distributed systems have changed the face of the world. When your web browser connects to a web server somewhere else on the planet, it is participating in what seems to be a simple form of a **client/server** distributed system. When you contact a modern web service such as Google or Facebook, you are not just interacting with a single machine, however; behind the scenes, these complex services are built from a large collection (i.e., thousands) of machines, each of which cooperate to provide the particular service of the site. Thus, it should be clear what makes studying distributed systems interesting. Indeed, it is worthy of an entire class; here, we just introduce a few of the major topics.

A number of new challenges arise when building a distributed system. The major one we focus on is **failure**; machines, disks, networks, and software all fail from time to time, as we do not (and likely, will never) know how to build "perfect" components and systems. However, when we build a modern web service, we'd like it to appear to clients as if it never fails; how can we accomplish this task?

THE CRUX:
HOW TO BUILD SYSTEMS THAT WORK WHEN COMPONENTS FAIL
How can we build a working system out of parts that don't work correctly all the time? The basic question should remind you of some of the topics we discussed in RAID storage arrays; however, the problems here tend to be more complex, as are the solutions.

Interestingly, while failure is a central challenge in constructing distributed systems, it also represents an opportunity. Yes, machines fail; but the mere fact that a machine fails does not imply the entire system must fail. By collecting together a set of machines, we can build a system that appears to rarely fail, despite the fact that its components fail regularly. This reality is the central beauty and value of distributed systems, and why they underly virtually every modern web service you use, including Google, Facebook, etc.

TIP: COMMUNICATION IS INHERENTLY UNRELIABLE
In virtually all circumstances, it is good to view communication as a fundamentally unreliable activity. Bit corruption, down or non-working links and machines, and lack of buffer space for incoming packets all lead to the same result: packets sometimes do not reach their destination. To build reliable services atop such unreliable networks, we must consider techniques that can cope with packet loss.

Other important issues exist as well. System **performance** is often critical; with a network connecting our distributed system together, system designers must often think carefully about how to accomplish their given tasks, trying to reduce the number of messages sent and further make communication as efficient (low latency, high bandwidth) as possible.

Finally, **security** is also a necessary consideration. When connecting to a remote site, having some assurance that the remote party is who they say they are becomes a central problem. Further, ensuring that third parties cannot monitor or alter an on-going communication between two others is also a challenge.

In this introduction, we'll cover the most basic new aspect that is new in a distributed system: **communication**. Namely, how should machines within a distributed system communicate with one another? We'll start with the most basic primitives available, messages, and build a few higher-level primitives on top of them. As we said above, failure will be a central focus: how should communication layers handle failures?

47.1 Communication Basics

The central tenet of modern networking is that communication is fundamentally unreliable. Whether in the wide-area Internet, or a local-area high-speed network such as Infiniband, packets are regularly lost, corrupted, or otherwise do not reach their destination.

There are a multitude of causes for packet loss or corruption. Sometimes, during transmission, some bits get flipped due to electrical or other similar problems. Sometimes, an element in the system, such as a network link or packet router or even the remote host, are somehow damaged or otherwise not working correctly; network cables do accidentally get severed, at least sometimes.

More fundamental however is packet loss due to lack of buffering within a network switch, router, or endpoint. Specifically, even if we could guarantee that all links worked correctly, and that all the components in the system (switches, routers, end hosts) were up and running as expected, loss is still possible, for the following reason. Imagine a packet arrives at a router; for the packet to be processed, it must be placed in memory somewhere within the router. If many such packets arrive at

```
// client code
int main(int argc, char *argv[]) {
    int sd = UDP_Open(20000);
    struct sockaddr_in addrSnd, addrRcv;
    int rc = UDP_FillSockAddr(&addrSnd, "machine.cs.wisc.edu", 10000);
    char message[BUFFER_SIZE];
    sprintf(message, "hello world");
    rc = UDP_Write(sd, &addrSnd, message, BUFFER_SIZE);
    if (rc > 0) {
        int rc = UDP_Read(sd, &addrRcv, message, BUFFER_SIZE);
    }
    return 0;
}

// server code
int main(int argc, char *argv[]) {
    int sd = UDP_Open(10000);
    assert(sd > -1);
    while (1) {
        struct sockaddr_in addr;
        char message[BUFFER_SIZE];
        int rc = UDP_Read(sd, &addr, message, BUFFER_SIZE);
        if (rc > 0) {
            char reply[BUFFER_SIZE];
            sprintf(reply, "goodbye world");
            rc = UDP_Write(sd, &addr, reply, BUFFER_SIZE);
        }
    }
    return 0;
}
```

Figure 47.1: **Example UDP/IP Client/Server Code**

once, it is possible that the memory within the router cannot accommodate all of the packets. The only choice the router has at that point is to **drop** one or more of the packets. This same behavior occurs at end hosts as well; when you send a large number of messages to a single machine, the machine's resources can easily become overwhelmed, and thus packet loss again arises.

Thus, packet loss is fundamental in networking. The question thus becomes: how should we deal with it?

47.2 Unreliable Communication Layers

One simple way is this: we don't deal with it. Because some applications know how to deal with packet loss, it is sometimes useful to let them communicate with a basic unreliable messaging layer, an example of the **end-to-end argument** one often hears about (see the **Aside** at end of chapter). One excellent example of such an unreliable layer is found in the **UDP/IP** networking stack available today on virtually all modern systems. To use UDP, a process uses the **sockets** API in order to create a **communication endpoint**; processes on other machines (or on the same machine) send UDP **datagrams** to the original process (a datagram is a fixed-sized message up to some max size).

```
int UDP_Open(int port) {
    int sd;
    if ((sd = socket(AF_INET, SOCK_DGRAM, 0)) == -1) { return -1; }
    struct sockaddr_in myaddr;
    bzero(&myaddr, sizeof(myaddr));
    myaddr.sin_family      = AF_INET;
    myaddr.sin_port        = htons(port);
    myaddr.sin_addr.s_addr = INADDR_ANY;
    if (bind(sd, (struct sockaddr *) &myaddr, sizeof(myaddr)) == -1) {
        close(sd);
        return -1;
    }
    return sd;
}

int UDP_FillSockAddr(struct sockaddr_in *addr, char *hostName, int port) {
    bzero(addr, sizeof(struct sockaddr_in));
    addr->sin_family = AF_INET;          // host byte order
    addr->sin_port   = htons(port);      // short, network byte order
    struct in_addr *inAddr;
    struct hostent *hostEntry;
    if ((hostEntry = gethostbyname(hostName)) == NULL) { return -1; }
    inAddr = (struct in_addr *) hostEntry->h_addr;
    addr->sin_addr = *inAddr;
    return 0;
}

int UDP_Write(int sd, struct sockaddr_in *addr, char *buffer, int n) {
    int addrLen = sizeof(struct sockaddr_in);
    return sendto(sd, buffer, n, 0, (struct sockaddr *) addr, addrLen);
}

int UDP_Read(int sd, struct sockaddr_in *addr, char *buffer, int n) {
    int len = sizeof(struct sockaddr_in);
    return recvfrom(sd, buffer, n, 0, (struct sockaddr *) addr,
                    (socklen_t *) &len);
}
```

Figure 47.2: **A Simple UDP Library**

Figures 47.1 and 47.2 show a simple client and server built on top of UDP/IP. The client can send a message to the server, which then responds with a reply. With this small amount of code, you have all you need to begin building distributed systems!

UDP is a great example of an unreliable communication layer. If you use it, you will encounter situations where packets get lost (dropped) and thus do not reach their destination; the sender is never thus informed of the loss. However, that does not mean that UDP does not guard against any failures at all. For example, UDP includes a **checksum** to detect some forms of packet corruption.

However, because many applications simply want to send data to a destination and not worry about packet loss, we need more. Specifically, we need reliable communication on top of an unreliable network.

TIP: USE CHECKSUMS FOR INTEGRITY

Checksums are a commonly-used method to detect corruption quickly and effectively in modern systems. A simple checksum is addition: just sum up the bytes of a chunk of data; of course, many other more sophisticated checksums have been created, including basic cyclic redundancy codes (CRCs), the Fletcher checksum, and many others [MK09].

In networking, checksums are used as follows. Before sending a message from one machine to another, compute a checksum over the bytes of the message. Then send both the message and the checksum to the destination. At the destination, the receiver computes a checksum over the incoming message as well; if this computed checksum matches the sent checksum, the receiver can feel some assurance that the data likely did not get corrupted during transmission.

Checksums can be evaluated along a number of different axes. Effectiveness is one primary consideration: does a change in the data lead to a change in the checksum? The stronger the checksum, the harder it is for changes in the data to go unnoticed. Performance is the other important criterion: how costly is the checksum to compute? Unfortunately, effectiveness and performance are often at odds, meaning that checksums of high quality are often expensive to compute. Life, again, isn't perfect.

47.3 Reliable Communication Layers

To build a reliable communication layer, we need some new mechanisms and techniques to handle packet loss. Let us consider a simple example in which a client is sending a message to a server over an unreliable connection. The first question we must answer: how does the sender know that the receiver has actually received the message?

The technique that we will use is known as an **acknowledgment**, or **ack** for short. The idea is simple: the sender sends a message to the receiver; the receiver then sends a short message back to *acknowledge* its receipt. Figure 47.3 depicts the process.

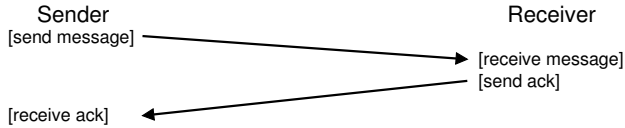


Figure 47.3: **Message Plus Acknowledgment**

When the sender receives an acknowledgment of the message, it can then rest assured that the receiver did indeed receive the original message. However, what should the sender do if it does not receive an acknowledgment?

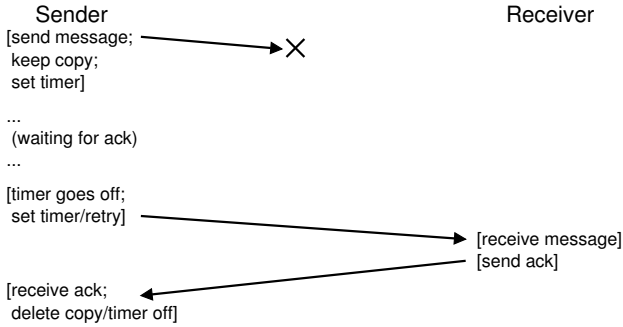


Figure 47.4: **Message Plus Acknowledgment: Dropped Request**

To handle this case, we need an additional mechanism, known as a **timeout**. When the sender sends a message, the sender now sets a timer to go off after some period of time. If, in that time, no acknowledgment has been received, the sender concludes that the message has been lost. The sender then simply performs a **retry** of the send, sending the same message again with hopes that this time, it will get through. For this approach to work, the sender must keep a copy of the message around, in case it needs to send it again. The combination of the timeout and the retry have led some to call the approach **timeout/retry**; pretty clever crowd, those networking types, no? Figure 47.4 shows an example.

Unfortunately, timeout/retry in this form is not quite enough. Figure 47.5 shows an example of packet loss which could lead to trouble. In this example, it is not the original message that gets lost, but the acknowledgment. From the perspective of the sender, the situation seems the same: no ack was received, and thus a timeout and retry are in order. But from the perspective of the receiver, it is quite different: now the same message has been received twice! While there may be cases where this is OK, in general it is not; imagine what would happen when you are downloading a file and extra packets are repeated inside the download. Thus, when we are aiming for a reliable message layer, we also usually want to guarantee that each message is received **exactly once** by the receiver.

To enable the receiver to detect duplicate message transmission, the sender has to identify each message in some unique way, and the receiver needs some way to track whether it has already seen each message before. When the receiver sees a duplicate transmission, it simply acks the message, but (critically) does *not* pass the message to the application that receives the data. Thus, the sender receives the ack but the message is not received twice, preserving the exactly-once semantics mentioned above.

There are myriad ways to detect duplicate messages. For example, the sender could generate a unique ID for each message; the receiver could track every ID it has ever seen. This approach could work, but it is prohibitively costly, requiring unbounded memory to track all IDs.

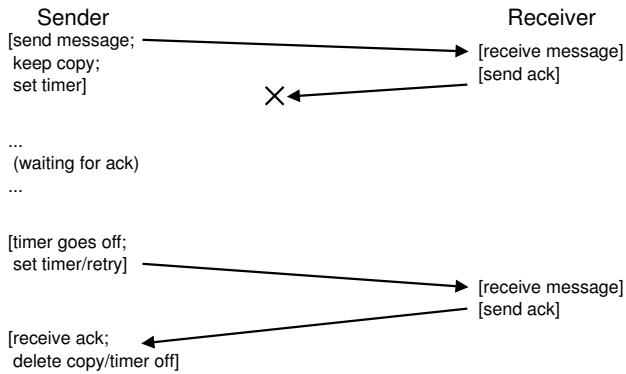


Figure 47.5: **Message Plus Acknowledgment: Dropped Reply**

A simpler approach, requiring little memory, solves this problem, and the mechanism is known as a **sequence counter**. With a sequence counter, the sender and receiver agree upon a start value (e.g., 1) for a counter that each side will maintain. Whenever a message is sent, the current value of the counter is sent along with the message; this counter value (N) serves as an ID for the message. After the message is sent, the sender then increments the value (to $N + 1$).

The receiver uses its counter value as the expected value for the ID of the incoming message from that sender. If the ID of a received message (N) matches the receiver's counter (also N), it acks the message and passes it up to the application; in this case, the receiver concludes this is the first time this message has been received. The receiver then increments its counter (to $N + 1$), and waits for the next message.

If the ack is lost, the sender will timeout and re-send message N. This time, the receiver's counter is higher ($N+1$), and thus the receiver knows it has already received this message. Thus it acks the message but does *not* pass it up to the application. In this simple manner, sequence counters can be used to avoid duplicates.

The most commonly used reliable communication layer is known as **TCP/IP**, or just **TCP** for short. TCP has a great deal more sophistication than we describe above, including machinery to handle congestion in the network [VJ88], multiple outstanding requests, and hundreds of other small tweaks and optimizations. Read more about it if you're curious; better yet, take a networking course and learn that material well.

47.4 Communication Abstractions

Given a basic messaging layer, we now approach the next question in this chapter: what abstraction of communication should we use when building a distributed system?

TIP: BE CAREFUL SETTING THE TIMEOUT VALUE

As you can probably guess from the discussion, setting the timeout value correctly is an important aspect of using timeouts to retry message sends. If the timeout is too small, the sender will re-send messages needlessly, thus wasting CPU time on the sender and network resources. If the timeout is too large, the sender waits too long to re-send and thus perceived performance at the sender is reduced. The "right" value, from the perspective of a single client and server, is thus to wait just long enough to detect packet loss but no longer.

However, there are often more than just a single client and server in a distributed system, as we will see in future chapters. In a scenario with many clients sending to a single server, packet loss at the server may be an indicator that the server is overloaded. If true, clients might retry in a different adaptive manner; for example, after the first timeout, a client might increase its timeout value to a higher amount, perhaps twice as high as the original value. Such an **exponential back-off** scheme, pioneered in the early Aloha network and adopted in early Ethernet [A70], avoids situations where resources are being overloaded by an excess of re-sends. Robust systems strive to avoid overload of this nature.

The systems community developed a number of approaches over the years. One body of work took OS abstractions and extended them to operate in a distributed environment. For example, **distributed shared memory** (**DSM**) systems enable processes on different machines to share a large, virtual address space [LH89]. This abstraction turns a distributed computation into something that looks like a multi-threaded application; the only difference is that these threads run on different machines instead of different processors within the same machine.

The way most DSM systems work is through the virtual memory system of the OS. When a page is accessed on one machine, two things can happen. In the first (best) case, the page is already local on the machine, and thus the data is fetched quickly. In the second case, the page is currently on some other machine. A page fault occurs, and the page fault handler sends a message to some other machine to fetch the page, install it in the page table of the requesting process, and continue execution.

This approach is not widely in use today for a number of reasons. The largest problem for DSM is how it handles failure. Imagine, for example, if a machine fails; what happens to the pages on that machine? What if the data structures of the distributed computation are spread across the entire address space? In this case, parts of these data structures would suddenly become unavailable. Dealing with failure when parts of your address space go missing is hard; imagine a linked list that where a next pointer points into a portion of the address space that is gone. Yikes!

A further problem is performance. One usually assumes, when writing code, that access to memory is cheap. In DSM systems, some accesses

are inexpensive, but others cause page faults and expensive fetches from remote machines. Thus, programmers of such DSM systems had to be very careful to organize computations such that almost no communication occurred at all, defeating much of the point of such an approach. Though much research was performed in this space, there was little practical impact; nobody builds reliable distributed systems using DSM today.

47.5 Remote Procedure Call (RPC)

While OS abstractions turned out to be a poor choice for building distributed systems, programming language (PL) abstractions make much more sense. The most dominant abstraction is based on the idea of a **remote procedure call**, or **RPC** for short [BN84][1].

Remote procedure call packages all have a simple goal: to make the process of executing code on a remote machine as simple and straightforward as calling a local function. Thus, to a client, a procedure call is made, and some time later, the results are returned. The server simply defines some routines that it wishes to export. The rest of the magic is handled by the RPC system, which in general has two pieces: a **stub generator** (sometimes called a **protocol compiler**), and the **run-time library**. We'll now take a look at each of these pieces in more detail.

Stub Generator

The stub generator's job is simple: to remove some of the pain of packing function arguments and results into messages by automating it. Numerous benefits arise: one avoids, by design, the simple mistakes that occur in writing such code by hand; further, a stub compiler can perhaps optimize such code and thus improve performance.

The input to such a compiler is simply the set of calls a server wishes to export to clients. Conceptually, it could be something as simple as this:

```
interface {
  int func1(int arg1);
  int func2(int arg1, int arg2);
};
```

The stub generator takes an interface like this and generates a few different pieces of code. For the client, a **client stub** is generated, which contains each of the functions specified in the interface; a client program wishing to use this RPC service would link with this client stub and call into it in order to make RPCs.

Internally, each of these functions in the client stub do all of the work needed to perform the remote procedure call. To the client, the code just

[1] In modern programming languages, we might instead say **remote method invocation** (**RMI**), but who likes these languages anyhow, with all of their fancy objects?

appears as a function call (e.g., the client calls `func1(x)`); internally, the code in the client stub for `func1()` does this:

- **Create a message buffer.** A message buffer is usually just a contiguous array of bytes of some size.
- **Pack the needed information into the message buffer.** This information includes some kind of identifier for the function to be called, as well as all of the arguments that the function needs (e.g., in our example above, one integer for `func1`). The process of putting all of this information into a single contiguous buffer is sometimes referred to as the **marshaling** of arguments or the **serialization** of the message.
- **Send the message to the destination RPC server.** The communication with the RPC server, and all of the details required to make it operate correctly, are handled by the RPC run-time library, described further below.
- **Wait for the reply.** Because function calls are usually **synchronous**, the call will wait for its completion.
- **Unpack return code and other arguments.** If the function just returns a single return code, this process is straightforward; however, more complex functions might return more complex results (e.g., a list), and thus the stub might need to unpack those as well. This step is also known as **unmarshaling** or **deserialization**.
- **Return to the caller.** Finally, just return from the client stub back into the client code.

For the server, code is also generated. The steps taken on the server are as follows:

- **Unpack the message.** This step, called **unmarshaling** or **deserialization**, takes the information out of the incoming message. The function identifier and arguments are extracted.
- **Call into the actual function.** Finally! We have reached the point where the remote function is actually executed. The RPC runtime calls into the function specified by the ID and passes in the desired arguments.
- **Package the results.** The return argument(s) are marshaled back into a single reply buffer.
- **Send the reply.** The reply is finally sent to the caller.

There are a few other important issues to consider in a stub compiler. The first is complex arguments, i.e., how does one package and send a complex data structure? For example, when one calls the `write()` system call, one passes in three arguments: an integer file descriptor, a pointer to a buffer, and a size indicating how many bytes (starting at the pointer) are to be written. If an RPC package is passed a pointer, it needs to be able to figure out how to interpret that pointer, and perform the

correct action. Usually this is accomplished through either well-known types (e.g., a `buffer_t` that is used to pass chunks of data given a size, which the RPC compiler understands), or by annotating the data structures with more information, enabling the compiler to know which bytes need to be serialized.

Another important issue is the organization of the server with regards to concurrency. A simple server just waits for requests in a simple loop, and handles each request one at a time. However, as you might have guessed, this can be grossly inefficient; if one RPC call blocks (e.g., on I/O), server resources are wasted. Thus, most servers are constructed in some sort of concurrent fashion. A common organization is a **thread pool**. In this organization, a finite set of threads are created when the server starts; when a message arrives, it is dispatched to one of these worker threads, which then does the work of the RPC call, eventually replying; during this time, a main thread keeps receiving other requests, and perhaps dispatching them to other workers. Such an organization enables concurrent execution within the server, thus increasing its utilization; the standard costs arise as well, mostly in programming complexity, as the RPC calls may now need to use locks and other synchronization primitives in order to ensure their correct operation.

Run-Time Library

The run-time library handles much of the heavy lifting in an RPC system; most performance and reliability issues are handled herein. We'll now discuss some of the major challenges in building such a run-time layer.

One of the first challenges we must overcome is how to locate a remote service. This problem, of **naming**, is a common one in distributed systems, and in some sense goes beyond the scope of our current discussion. The simplest of approaches build on existing naming systems, e.g., hostnames and port numbers provided by current internet protocols. In such a system, the client must know the hostname or IP address of the machine running the desired RPC service, as well as the port number it is using (a port number is just a way of identifying a particular communication activity taking place on a machine, allowing multiple communication channels at once). The protocol suite must then provide a mechanism to route packets to a particular address from any other machine in the system. For a good discussion of naming, you'll have to look elsewhere, e.g., read about DNS and name resolution on the Internet, or better yet just read the excellent chapter in Saltzer and Kaashoek's book [SK09].

Once a client knows which server it should talk to for a particular remote service, the next question is which transport-level protocol should RPC be built upon. Specifically, should the RPC system use a reliable protocol such as TCP/IP, or be built upon an unreliable communication layer such as UDP/IP?

Naively the choice would seem easy: clearly we would like for a request to be reliably delivered to the remote server, and clearly we would

like to reliably receive a reply. Thus we should choose the reliable transport protocol such as TCP, right?

Unfortunately, building RPC on top of a reliable communication layer can lead to a major inefficiency in performance. Recall from the discussion above how reliable communication layers work: with acknowledgments plus timeout/retry. Thus, when the client sends an RPC request to the server, the server responds with an acknowledgment so that the caller knows the request was received. Similarly, when the server sends the reply to the client, the client acks it so that the server knows it was received. By building a request/response protocol (such as RPC) on top of a reliable communication layer, two "extra" messages are sent.

For this reason, many RPC packages are built on top of unreliable communication layers, such as UDP. Doing so enables a more efficient RPC layer, but does add the responsibility of providing reliability to the RPC system. The RPC layer achieves the desired level of responsibility by using timeout/retry and acknowledgments much like we described above. By using some form of sequence numbering, the communication layer can guarantee that each RPC takes place exactly once (in the case of no failure), or at most once (in the case where failure arises).

Other Issues

There are some other issues an RPC run-time must handle as well. For example, what happens when a remote call takes a long time to complete? Given our timeout machinery, a long-running remote call might appear as a failure to a client, thus triggering a retry, and thus the need for some care here. One solution is to use an explicit acknowledgment (from the receiver to sender) when the reply isn't immediately generated; this lets the client know the server received the request. Then, after some time has passed, the client can periodically ask whether the server is still working on the request; if the server keeps saying "yes", the client should be happy and continue to wait (after all, sometimes a procedure call can take a long time to finish executing).

The run-time must also handle procedure calls with large arguments, larger than what can fit into a single packet. Some lower-level network protocols provide such sender-side **fragmentation** (of larger packets into a set of smaller ones) and receiver-side **reassembly** (of smaller parts into one larger logical whole); if not, the RPC run-time may have to implement such functionality itself. See Birrell and Nelson's excellent RPC paper for details [BN84].

One issue that many systems handle is that of **byte ordering**. As you may know, some machines store values in what is known as **big endian** ordering, whereas others use **little endian** ordering. Big endian stores bytes (say, of an integer) from most significant to least significant bits, much like Arabic numerals; little endian does the opposite. Both are equally valid ways of storing numeric information; the question here is how to communicate between machines of *different* endianness.

Aside: **The End-to-End Argument**

The **end-to-end argument** makes the case that the highest level in a system, i.e., usually the application at "the end", is ultimately the only locale within a layered system where certain functionality can truly be implemented. In their landmark paper [SRC84], Saltzer et al. argue this through an excellent example: reliable file transfer between two machines. If you want to transfer a file from machine A to machine B, and make sure that the bytes that end up on B are exactly the same as those that began on A, you must have an "end-to-end" check of this; lower-level reliable machinery, e.g., in the network or disk, provides no such guarantee.

The contrast is an approach which tries to solve the reliable-file-transfer problem by adding reliability to lower layers of the system. For example, say we build a reliable communication protocol and use it to build our reliable file transfer. The communication protocol guarantees that every byte sent by a sender will be received in order by the receiver, say using timeout/retry, acknowledgments, and sequence numbers. Unfortunately, using such a protocol does not a reliable file transfer make; imagine the bytes getting corrupted in sender memory before the communication even takes place, or something bad happening when the receiver writes the data to disk. In those cases, even though the bytes were delivered reliably across the network, our file transfer was ultimately not reliable. To build a reliable file transfer, one must include end-to-end checks of reliability, e.g., after the entire transfer is complete, read back the file on the receiver disk, compute a checksum, and compare that checksum to that of the file on the sender.

The corollary to this maxim is that sometimes having lower layers provide extra functionality can indeed improve system performance or otherwise optimize a system. Thus, you should not rule out having such machinery at a lower-level in a system; rather, you should carefully consider the utility of such machinery, given its eventual usage in an overall system or application.

RPC packages often handle this by providing a well-defined endianness within their message formats. In Sun's RPC package, the **XDR** (**eXternal Data Representation**) layer provides this functionality. If the machine sending or receiving a message matches the endianness of XDR, messages are just sent and received as expected. If, however, the machine communicating has a different endianness, each piece of information in the message must be converted. Thus, the difference in endianness can have a small performance cost.

A final issue is whether to expose the asynchronous nature of communication to clients, thus enabling some performance optimizations. Specifically, typical RPCs are made **synchronously**, i.e., when a client issues the procedure call, it must wait for the procedure call to return

before continuing. Because this wait can be long, and because the client may have other work it could be doing, some RPC packages enable you to invoke an RPC **asynchronously**. When an asynchronous RPC is issued, the RPC package sends the request and returns immediately; the client is then free to do other work, such as call other RPCs or other useful computation. The client at some point will want to see the results of the asynchronous RPC; it thus calls back into the RPC layer, telling it to wait for outstanding RPCs to complete, at which point return arguments can be accessed.

47.6 Summary

We have seen the introduction of a new topic, distributed systems, and its major issue: how to handle failure which is now a commonplace event. As they say inside of Google, when you have just your desktop machine, failure is rare; when you're in a data center with thousands of machines, failure is happening all the time. The key to any distributed system is how you deal with that failure.

We have also seen that communication forms the heart of any distributed system. A common abstraction of that communication is found in remote procedure call (RPC), which enables clients to make remote calls on servers; the RPC package handles all of the gory details, including timeout/retry and acknowledgment, in order to deliver a service that closely mirrors a local procedure call.

The best way to really understand an RPC package is of course to use one yourself. Sun's RPC system, using the stub compiler `rpcgen`, is a common one, and is widely available on systems today, including Linux. Try it out, and see what all the fuss is about.

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[LH89] "Memory Coherence in Shared Virtual Memory Systems"
Kai Li and Paul Hudak
ACM TOCS, 7:4, November 1989
The introduction of software-based shared memory via virtual memory. An intriguing idea for sure, but not a lasting or good one in the end.

[SK09] "Principles of Computer System Design"
Jerome H. Saltzer and M. Frans Kaashoek
Morgan-Kaufmann, 2009
An excellent book on systems, and a must for every bookshelf. One of the few terrific discussions on naming we've seen.

[SRC84] "End-To-End Arguments in System Design"
Jerome H. Saltzer, David P. Reed, David D. Clark
ACM TOCS, 2:4, November 1984
A beautiful discussion of layering, abstraction, and where functionality must ultimately reside in computer systems.

[VJ88] "Congestion Avoidance and Control"
Van Jacobson
SIGCOMM '88
A pioneering paper on how clients should adjust to perceived network congestion; definitely one of the key pieces of technology underlying the Internet, and a must read for anyone serious about systems, and for Van Jacobson's relatives because well relatives should read all of your papers.

48

Sun's Network File System (NFS)

One of the first uses of distributed client/server computing was in the realm of distributed file systems. In such an environment, there are a number of client machines and one server (or a few); the server stores the data on its disks, and clients request data through well-formed protocol messages. Figure 48.1 depicts the basic setup.

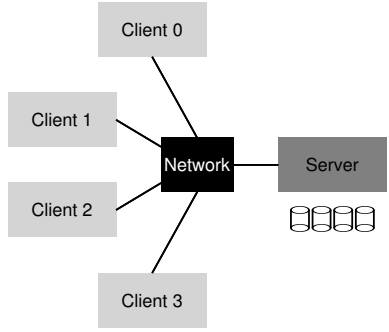

Figure 48.1: **A Generic Client/Server System**

As you can see from the picture, the server has the disks, and clients send messages across a network to access their directories and files on those disks. Why do we bother with this arrangement? (i.e., why don't we just let clients use their local disks?) Well, primarily this setup allows for easy **sharing** of data across clients. Thus, if you access a file on one machine (Client 0) and then later use another (Client 2), you will have the same view of the file system. Your data is naturally shared across these different machines. A secondary benefit is **centralized administration**; for example, backing up files can be done from the few server machines instead of from the multitude of clients. Another advantage could be **security**; having all servers in a locked machine room prevents certain types of problems from arising.

CRUX: HOW TO BUILD A DISTRIBUTED FILE SYSTEM
How do you build a distributed file system? What are the key aspects to think about? What is easy to get wrong? What can we learn from existing systems?

48.1 A Basic Distributed File System

We now will study the architecture of a simplified distributed file system. A simple client/server distributed file system has more components than the file systems we have studied so far. On the client side, there are client applications which access files and directories through the **client-side file system**. A client application issues **system calls** to the client-side file system (such as `open()`, `read()`, `write()`, `close()`, `mkdir()`, etc.) in order to access files which are stored on the server. Thus, to client applications, the file system does not appear to be any different than a local (disk-based) file system, except perhaps for performance; in this way, distributed file systems provide **transparent** access to files, an obvious goal; after all, who would want to use a file system that required a different set of APIs or otherwise was a pain to use?

The role of the client-side file system is to execute the actions needed to service those system calls. For example, if the client issues a `read()` request, the client-side file system may send a message to the **server-side file system** (or, as it is commonly called, the **file server**) to read a particular block; the file server will then read the block from disk (or its own in-memory cache), and send a message back to the client with the requested data. The client-side file system will then copy the data into the user buffer supplied to the `read()` system call and thus the request will complete. Note that a subsequent `read()` of the same block on the client may be **cached** in client memory or on the client's disk even; in the best such case, no network traffic need be generated.

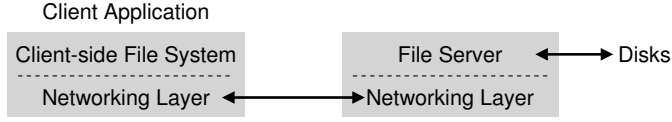

Figure 48.2: **Distributed File System Architecture**

From this simple overview, you should get a sense that there are two important pieces of software in a client/server distributed file system: the client-side file system and the file server. Together their behavior determines the behavior of the distributed file system. Now it's time to study one particular system: Sun's Network File System (NFS).

ASIDE: **WHY SERVERS CRASH**

Before getting into the details of the NFSv2 protocol, you might be wondering: why do servers crash? Well, as you might guess, there are plenty of reasons. Servers may simply suffer from a **power outage** (temporarily); only when power is restored can the machines be restarted. Servers are often comprised of hundreds of thousands or even millions of lines of code; thus, they have **bugs** (even good software has a few bugs per hundred or thousand lines of code), and thus they eventually will trigger a bug that will cause them to crash. They also have memory leaks; even a small memory leak will cause a system to run out of memory and crash. And, finally, in distributed systems, there is a network between the client and the server; if the network acts strangely (for example, if it becomes **partitioned** and clients and servers are working but cannot communicate), it may appear as if a remote machine has crashed, but in reality it is just not currently reachable through the network.

48.2 On To NFS

One of the earliest and quite successful distributed systems was developed by Sun Microsystems, and is known as the Sun Network File System (or NFS) [S86]. In defining NFS, Sun took an unusual approach: instead of building a proprietary and closed system, Sun instead developed an **open protocol** which simply specified the exact message formats that clients and servers would use to communicate. Different groups could develop their own NFS servers and thus compete in an NFS marketplace while preserving interoperability. It worked: today there are many companies that sell NFS servers (including Oracle/Sun, NetApp [HLM94], EMC, IBM, and others), and the widespread success of NFS is likely attributed to this "open market" approach.

48.3 Focus: Simple and Fast Server Crash Recovery

In this chapter, we will discuss the classic NFS protocol (version 2, a.k.a. NFSv2), which was the standard for many years; small changes were made in moving to NFSv3, and larger-scale protocol changes were made in moving to NFSv4. However, NFSv2 is both wonderful and frustrating and thus serves as our focus.

In NFSv2, the main goal in the design of the protocol was *simple and fast server crash recovery*. In a multiple-client, single-server environment, this goal makes a great deal of sense; any minute that the server is down (or unavailable) makes *all* the client machines (and their users) unhappy and unproductive. Thus, as the server goes, so goes the entire system.

48.4 Key To Fast Crash Recovery: Statelessness

This simple goal is realized in NFSv2 by designing what we refer to as a **stateless** protocol. The server, by design, does not keep track of anything about what is happening at each client. For example, the server does not know which clients are caching which blocks, or which files are currently open at each client, or the current file pointer position for a file, etc. Simply put, the server does not track anything about what clients are doing; rather, the protocol is designed to deliver in each protocol request *all the information* that is needed in order to complete the request. If it doesn't now, this stateless approach will make more sense as we discuss the protocol in more detail below.

For an example of a **stateful** (not stateless) protocol, consider the `open()` system call. Given a pathname, `open()` returns a file descriptor (an integer). This descriptor is used on subsequent `read()` or `write()` requests to access various file blocks, as in this application code (note that proper error checking of the system calls is omitted for space reasons):

```
char buffer[MAX];
int fd = open("foo", O_RDONLY); // get descriptor "fd"
read(fd, buffer, MAX);          // read MAX bytes from foo (via fd)
read(fd, buffer, MAX);          // read MAX bytes from foo
...
read(fd, buffer, MAX);          // read MAX bytes from foo
close(fd);                      // close file
```

Figure 48.3: **Client Code: Reading From A File**

Now imagine that the client-side file system opens the file by sending a protocol message to the server saying "open the file 'foo' and give me back a descriptor". The file server then opens the file locally on its side and sends the descriptor back to the client. On subsequent reads, the client application uses that descriptor to call the `read()` system call; the client-side file system then passes the descriptor in a message to the file server, saying "read some bytes from the file that is referred to by the descriptor I am passing you here".

In this example, the file descriptor is a piece of **shared state** between the client and the server (Ousterhout calls this **distributed state** [O91]). Shared state, as we hinted above, complicates crash recovery. Imagine the server crashes after the first read completes, but before the client has issued the second one. After the server is up and running again, the client then issues the second read. Unfortunately, the server has no idea to which file `fd` is referring; that information was ephemeral (i.e., in memory) and thus lost when the server crashed. To handle this situation, the client and server would have to engage in some kind of **recovery protocol**, where the client would make sure to keep enough information around in its memory to be able to tell the server what it needs to know (in this case, that file descriptor `fd` refers to file `foo`).

It gets even worse when you consider the fact that a stateful server has to deal with client crashes. Imagine, for example, a client that opens a file and then crashes. The `open()` uses up a file descriptor on the server; how can the server know it is OK to close a given file? In normal operation, a client would eventually call `close()` and thus inform the server that the file should be closed. However, when a client crashes, the server never receives a `close()`, and thus has to notice the client has crashed in order to close the file.

For these reasons, the designers of NFS decided to pursue a stateless approach: each client operation contains all the information needed to complete the request. No fancy crash recovery is needed; the server just starts running again, and a client, at worst, might have to retry a request.

48.5 The NFSv2 Protocol

We thus arrive at the NFSv2 protocol definition. Our problem statement is simple:

> THE CRUX: HOW TO DEFINE A STATELESS FILE PROTOCOL
> How can we define the network protocol to enable stateless operation? Clearly, stateful calls like `open()` can't be a part of the discussion (as it would require the server to track open files); however, the client application will want to call `open()`, `read()`, `write()`, `close()` and other standard API calls to access files and directories. Thus, as a refined question, how do we define the protocol to both be stateless *and* support the POSIX file system API?

One key to understanding the design of the NFS protocol is understanding the **file handle**. File handles are used to uniquely describe the file or directory a particular operation is going to operate upon; thus, many of the protocol requests include a file handle.

You can think of a file handle as having three important components: a *volume identifier*, an *inode number*, and a *generation number*; together, these three items comprise a unique identifier for a file or directory that a client wishes to access. The volume identifier informs the server which file system the request refers to (an NFS server can export more than one file system); the inode number tells the server which file within that partition the request is accessing. Finally, the generation number is needed when reusing an inode number; by incrementing it whenever an inode number is reused, the server ensures that a client with an old file handle can't accidentally access the newly-allocated file.

Here is a summary of some of the important pieces of the protocol; the full protocol is available elsewhere (see Callaghan's book for an excellent and detailed overview of NFS [C00]).

```
NFSPROC_GETATTR
  expects: file handle
  returns: attributes
NFSPROC_SETATTR
  expects: file handle, attributes
  returns: nothing
NFSPROC_LOOKUP
  expects: directory file handle, name of file/directory to look up
  returns: file handle
NFSPROC_READ
  expects: file handle, offset, count
  returns: data, attributes
NFSPROC_WRITE
  expects: file handle, offset, count, data
  returns: attributes
NFSPROC_CREATE
  expects: directory file handle, name of file, attributes
  returns: nothing
NFSPROC_REMOVE
  expects: directory file handle, name of file to be removed
  returns: nothing
NFSPROC_MKDIR
  expects: directory file handle, name of directory, attributes
  returns: file handle
NFSPROC_RMDIR
  expects: directory file handle, name of directory to be removed
  returns: nothing
NFSPROC_READDIR
  expects: directory handle, count of bytes to read, cookie
  returns: directory entries, cookie (to get more entries)
```

Figure 48.4: **The NFS Protocol: Examples**

We briefly highlight the important components of the protocol. First, the LOOKUP protocol message is used to obtain a file handle, which is then subsequently used to access file data. The client passes a directory file handle and name of a file to look up, and the handle to that file (or directory) plus its attributes are passed back to the client from the server.

For example, assume the client already has a directory file handle for the root directory of a file system (`/`) (indeed, this would be obtained through the NFS **mount protocol**, which is how clients and servers first are connected together; we do not discuss the mount protocol here for sake of brevity). If an application running on the client opens the file `/foo.txt`, the client-side file system sends a lookup request to the server, passing it the root file handle and the name `foo.txt`; if successful, the file handle (and attributes) for `foo.txt` will be returned.

In case you are wondering, attributes are just the metadata that the file system tracks about each file, including fields such as file creation time, last modification time, size, ownership and permissions information, and so forth, i.e., the same type of information that you would get back if you called `stat()` on a file.

Once a file handle is available, the client can issue READ and WRITE protocol messages on a file to read or write the file, respectively. The READ protocol message requires the protocol to pass along the file handle

of the file along with the offset within the file and number of bytes to read. The server then will be able to issue the read (after all, the handle tells the server which volume and which inode to read from, and the offset and count tells it which bytes of the file to read) and return the data to the client (or an error if there was a failure). WRITE is handled similarly, except the data is passed from the client to the server, and just a success code is returned.

One last interesting protocol message is the GETATTR request; given a file handle, it simply fetches the attributes for that file, including the last modified time of the file. We will see why this protocol request is important in NFSv2 below when we discuss caching (can you guess why?).

48.6 From Protocol to Distributed File System

Hopefully you are now getting some sense of how this protocol is turned into a file system across the client-side file system and the file server. The client-side file system tracks open files, and generally translates application requests into the relevant set of protocol messages. The server simply responds to each protocol message, each of which has all the information needed to complete request.

For example, let us consider a simple application which reads a file. In the diagram (Figure 48.5), we show what system calls the application makes, and what the client-side file system and file server do in responding to such calls.

A few comments about the figure. First, notice how the client tracks all relevant **state** for the file access, including the mapping of the integer file descriptor to an NFS file handle as well as the current file pointer. This enables the client to turn each read request (which you may have noticed do *not* specify the offset to read from explicitly) into a properly-formatted read protocol message which tells the server exactly which bytes from the file to read. Upon a successful read, the client updates the current file position; subsequent reads are issued with the same file handle but a different offset.

Second, you may notice where server interactions occur. When the file is opened for the first time, the client-side file system sends a LOOKUP request message. Indeed, if a long pathname must be traversed (e.g., `/home/remzi/foo.txt`), the client would send three LOOKUPs: one to look up `home` in the directory `/`, one to look up `remzi` in `home`, and finally one to look up `foo.txt` in `remzi`.

Third, you may notice how each server request has all the information needed to complete the request in its entirety. This design point is critical to be able to gracefully recover from server failure, as we will now discuss in more detail; it ensures that the server does not need state to be able to respond to the request.

Client	Server
fd = open("/foo", ...);	
Send LOOKUP (rootdir FH, "foo")	
	Receive LOOKUP request look for "foo" in root dir return foo's FH + attributes
Receive LOOKUP reply allocate file desc in open file table store foo's FH in table store current file position (0) return file descriptor to application	
read(fd, buffer, MAX);	
Index into open file table with fd get NFS file handle (FH) use current file position as offset	
Send READ (FH, offset=0, count=MAX)	
	Receive READ request use FH to get volume/inode num read inode from disk (or cache) compute block location (using offset) read data from disk (or cache) return data to client
Receive READ reply update file position (+bytes read) set current file position = MAX return data/error code to app	
read(fd, buffer, MAX);	
Same except offset=MAX and set current file position = 2*MAX	
read(fd, buffer, MAX);	
Same except offset=2*MAX and set current file position = 3*MAX	
close(fd);	
Just need to clean up local structures Free descriptor "fd" in open file table (No need to talk to server)	

Figure 48.5: **Reading A File: Client-side And File Server Actions**

TIP: IDEMPOTENCY IS POWERFUL
Idempotency is a useful property when building reliable systems. When an operation can be issued more than once, it is much easier to handle failure of the operation; you can just retry it. If an operation is *not* idempotent, life becomes more difficult.

48.7 Handling Server Failure with Idempotent Operations

When a client sends a message to the server, it sometimes does not receive a reply. There are many possible reasons for this failure to respond. In some cases, the message may be dropped by the network; networks do lose messages, and thus either the request or the reply could be lost and thus the client would never receive a response.

It is also possible that the server has crashed, and thus is not currently responding to messages. After a bit, the server will be rebooted and start running again, but in the meanwhile all requests have been lost. In all of these cases, clients are left with a question: what should they do when the server does not reply in a timely manner?

In NFSv2, a client handles all of these failures in a single, uniform, and elegant way: it simply *retries* the request. Specifically, after sending the request, the client sets a timer to go off after a specified time period. If a reply is received before the timer goes off, the timer is canceled and all is well. If, however, the timer goes off *before* any reply is received, the client assumes the request has not been processed and resends it. If the server replies, all is well and the client has neatly handled the problem.

The ability of the client to simply retry the request (regardless of what caused the failure) is due to an important property of most NFS requests: they are **idempotent**. An operation is called idempotent when the effect of performing the operation multiple times is equivalent to the effect of performing the operating a single time. For example, if you store a value to a memory location three times, it is the same as doing so once; thus "store value to memory" is an idempotent operation. If, however, you increment a counter three times, it results in a different amount than doing so just once; thus, "increment counter" is not idempotent. More generally, any operation that just reads data is obviously idempotent; an operation that updates data must be more carefully considered to determine if it has this property.

The heart of the design of crash recovery in NFS is the idempotency of most common operations. LOOKUP and READ requests are trivially idempotent, as they only read information from the file server and do not update it. More interestingly, WRITE requests are also idempotent. If, for example, a WRITE fails, the client can simply retry it. The WRITE message contains the data, the count, and (importantly) the exact offset to write the data to. Thus, it can be repeated with the knowledge that the outcome of multiple writes is the same as the outcome of a single one.

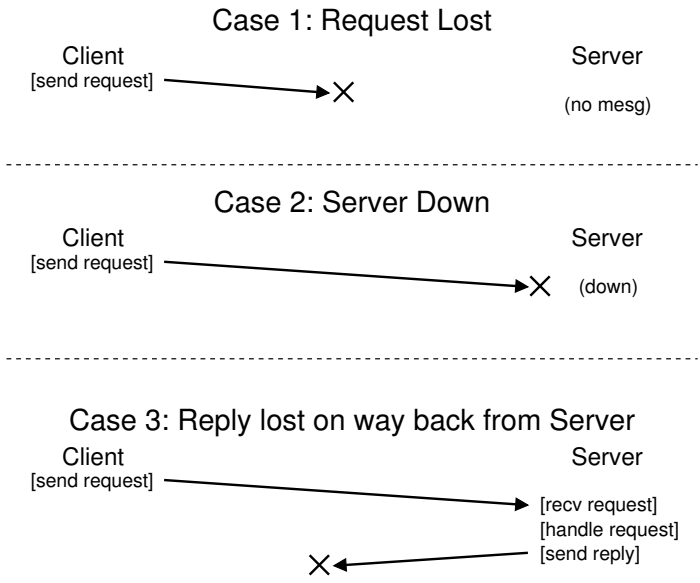

Figure 48.6: **The Three Types of Loss**

In this way, the client can handle all timeouts in a unified way. If a WRITE request was simply lost (Case 1 above), the client will retry it, the server will perform the write, and all will be well. The same will happen if the server happened to be down while the request was sent, but back up and running when the second request is sent, and again all works as desired (Case 2). Finally, the server may in fact receive the WRITE request, issue the write to its disk, and send a reply. This reply may get lost (Case 3), again causing the client to re-send the request. When the server receives the request again, it will simply do the exact same thing: write the data to disk and reply that it has done so. If the client this time receives the reply, all is again well, and thus the client has handled both message loss and server failure in a uniform manner. Neat!

A small aside: some operations are hard to make idempotent. For example, when you try to make a directory that already exists, you are informed that the mkdir request has failed. Thus, in NFS, if the file server receives a MKDIR protocol message and executes it successfully but the reply is lost, the client may repeat it and encounter that failure when in fact the operation at first succeeded and then only failed on the retry. Thus, life is not perfect.

TIP: PERFECT IS THE ENEMY OF THE GOOD (VOLTAIRE'S LAW)
Even when you design a beautiful system, sometimes all the corner cases don't work out exactly as you might like. Take the mkdir example above; one could redesign mkdir to have different semantics, thus making it idempotent (think about how you might do so); however, why bother? The NFS design philosophy covers most of the important cases, and overall makes the system design clean and simple with regards to failure. Thus, accepting that life isn't perfect and still building the system is a sign of good engineering. Apparently, this wisdom is attributed to Voltaire, for saying "... a wise Italian says that the best is the enemy of the good" [V72], and thus we call it **Voltaire's Law**.

48.8 Improving Performance: Client-side Caching

Distributed file systems are good for a number of reasons, but sending all read and write requests across the network can lead to a big performance problem: the network generally isn't that fast, especially as compared to local memory or disk. Thus, another problem: how can we improve the performance of a distributed file system?

The answer, as you might guess from reading the big bold words in the sub-heading above, is client-side **caching**. The NFS client-side file system caches file data (and metadata) that it has read from the server in client memory. Thus, while the first access is expensive (i.e., it requires network communication), subsequent accesses are serviced quite quickly out of client memory.

The cache also serves as a temporary buffer for writes. When a client application first writes to a file, the client buffers the data in client memory (in the same cache as the data it read from the file server) before writing the data out to the server. Such **write buffering** is useful because it decouples application `write()` latency from actual write performance, i.e., the application's call to `write()` succeeds immediately (and just puts the data in the client-side file system's cache); only later does the data get written out to the file server.

Thus, NFS clients cache data and performance is usually great and we are done, right? Unfortunately, not quite. Adding caching into any sort of system with multiple client caches introduces a big and interesting challenge which we will refer to as the **cache consistency problem**.

48.9 The Cache Consistency Problem

The cache consistency problem is best illustrated with two clients and a single server. Imagine client C1 reads a file F, and keeps a copy of the file in its local cache. Now imagine a different client, C2, overwrites the file F, thus changing its contents; let's call the new version of the file F

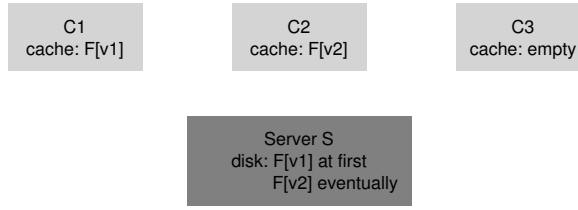

Figure 48.7: **The Cache Consistency Problem**

(version 2), or F[v2] and the old version F[v1] so we can keep the two distinct (but of course the file has the same name, just different contents). Finally, there is a third client, C3, which has not yet accessed the file F.

You can probably see the problem that is upcoming (Figure 48.7). In fact, there are two subproblems. The first subproblem is that the client C2 may buffer its writes in its cache for a time before propagating them to the server; in this case, while F[v2] sits in C2's memory, any access of F from another client (say C3) will fetch the old version of the file (F[v1]). Thus, by buffering writes at the client, other clients may get stale versions of the file, which may be undesirable; indeed, imagine the case where you log into machine C2, update F, and then log into C3 and try to read the file, only to get the old copy! Certainly this could be frustrating. Thus, let us call this aspect of the cache consistency problem **update visibility**; when do updates from one client become visible at other clients?

The second subproblem of cache consistency is a **stale cache**; in this case, C2 has finally flushed its writes to the file server, and thus the server has the latest version (F[v2]). However, C1 still has F[v1] in its cache; if a program running on C1 reads file F, it will get a stale version (F[v1]) and not the most recent copy (F[v2]), which is (often) undesirable.

NFSv2 implementations solve these cache consistency problems in two ways. First, to address update visibility, clients implement what is sometimes called **flush-on-close** (a.k.a., **close-to-open**) consistency semantics; specifically, when a file is written to and subsequently closed by a client application, the client flushes all updates (i.e., dirty pages in the cache) to the server. With flush-on-close consistency, NFS ensures that a subsequent open from another node will see the latest file version.

Second, to address the stale-cache problem, NFSv2 clients first check to see whether a file has changed before using its cached contents. Specifically, when opening a file, the client-side file system will issue a GETATTR request to the server to fetch the file's attributes. The attributes, importantly, include information as to when the file was last modified on the server; if the time-of-modification is more recent than the time that the file was fetched into the client cache, the client **invalidates** the file, thus removing it from the client cache and ensuring that subsequent reads will go to the server and retrieve the latest version of the file. If, on the other

hand, the client sees that it has the latest version of the file, it will go ahead and use the cached contents, thus increasing performance.

When the original team at Sun implemented this solution to the stale-cache problem, they realized a new problem; suddenly, the NFS server was flooded with GETATTR requests. A good engineering principle to follow is to design for the **common case**, and to make it work well; here, although the common case was that a file was accessed only from a single client (perhaps repeatedly), the client always had to send GETATTR requests to the server to make sure no one else had changed the file. A client thus bombards the server, constantly asking "has anyone changed this file?", when most of the time no one had.

To remedy this situation (somewhat), an **attribute cache** was added to each client. A client would still validate a file before accessing it, but most often would just look in the attribute cache to fetch the attributes. The attributes for a particular file were placed in the cache when the file was first accessed, and then would timeout after a certain amount of time (say 3 seconds). Thus, during those three seconds, all file accesses would determine that it was OK to use the cached file and thus do so with no network communication with the server.

48.10 Assessing NFS Cache Consistency

A few final words about NFS cache consistency. The flush-on-close behavior was added to "make sense", but introduced a certain performance problem. Specifically, if a temporary or short-lived file was created on a client and then soon deleted, it would still be forced to the server. A more ideal implementation might keep such short-lived files in memory until they are deleted and thus remove the server interaction entirely, perhaps increasing performance.

More importantly, the addition of an attribute cache into NFS made it very hard to understand or reason about exactly what version of a file one was getting. Sometimes you would get the latest version; sometimes you would get an old version simply because your attribute cache hadn't yet timed out and thus the client was happy to give you what was in client memory. Although this was fine most of the time, it would (and still does!) occasionally lead to odd behavior.

And thus we have described the oddity that is NFS client caching. It serves as an interesting example where details of an implementation serve to define user-observable semantics, instead of the other way around.

48.11 Implications on Server-Side Write Buffering

Our focus so far has been on client caching, and that is where most of the interesting issues arise. However, NFS servers tend to be well-equipped machines with a lot of memory too, and thus they have caching concerns as well. When data (and metadata) is read from disk, NFS

servers will keep it in memory, and subsequent reads of said data (and metadata) will not go to disk, a potential (small) boost in performance.

More intriguing is the case of write buffering. NFS servers absolutely may *not* return success on a WRITE protocol request until the write has been forced to stable storage (e.g., to disk or some other persistent device). While they can place a copy of the data in server memory, returning success to the client on a WRITE protocol request could result in incorrect behavior; can you figure out why?

The answer lies in our assumptions about how clients handle server failure. Imagine the following sequence of writes as issued by a client:

```
write(fd, a_buffer, size); // fill first block with a's
write(fd, b_buffer, size); // fill second block with b's
write(fd, c_buffer, size); // fill third block with c's
```

These writes overwrite the three blocks of a file with a block of a's, then b's, and then c's. Thus, if the file initially looked like this:

```
xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy
zzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzzz
```

We might expect the final result after these writes to be like this, with the x's, y's, and z's, would be overwritten with a's, b's, and c's, respectively.

```
aaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaa
bbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbbb
ccccccccccccccccccccccccccccccccccccccccccccccccccccccccccccccccc
```

Now let's assume for the sake of the example that these three client writes were issued to the server as three distinct WRITE protocol messages. Assume the first WRITE message is received by the server and issued to the disk, and the client informed of its success. Now assume the second write is just buffered in memory, and the server also reports it success to the client *before* forcing it to disk; unfortunately, the server crashes before writing it to disk. The server quickly restarts and receives the third write request, which also succeeds.

Thus, to the client, all the requests succeeded, but we are surprised that the file contents look like this:

```
aaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaa
yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy <--- oops
ccccccccccccccccccccccccccccccccccccccccccccccccccccccccccccccccc
```

Yikes! Because the server told the client that the second write was successful before committing it to disk, an old chunk is left in the file, which, depending on the application, might be catastrophic.

To avoid this problem, NFS servers *must* commit each write to stable (persistent) storage before informing the client of success; doing so enables the client to detect server failure during a write, and thus retry until

it finally succeeds. Doing so ensures we will never end up with file contents intermingled as in the above example.

The problem that this requirement gives rise to in NFS server implementation is that write performance, without great care, can be *the* major performance bottleneck. Indeed, some companies (e.g., Network Appliance) came into existence with the simple objective of building an NFS server that can perform writes quickly; one trick they use is to first put writes in a battery-backed memory, thus enabling to quickly reply to WRITE requests without fear of losing the data and without the cost of having to write to disk right away; the second trick is to use a file system design specifically designed to write to disk quickly when one finally needs to do so [HLM94, RO91].

48.12 Summary

We have seen the introduction of the NFS distributed file system. NFS is centered around the idea of simple and fast recovery in the face of server failure, and achieves this end through careful protocol design. Idempotency of operations is essential; because a client can safely replay a failed operation, it is OK to do so whether or not the server has executed the request.

We also have seen how the introduction of caching into a multiple-client, single-server system can complicate things. In particular, the system must resolve the cache consistency problem in order to behave reasonably; however, NFS does so in a slightly ad hoc fashion which can occasionally result in observably weird behavior. Finally, we saw how server caching can be tricky: writes to the server must be forced to stable storage before returning success (otherwise data can be lost).

We haven't talked about other issues which are certainly relevant, notably security. Security in early NFS implementations was remarkably lax; it was rather easy for any user on a client to masquerade as other users and thus gain access to virtually any file. Subsequent integration with more serious authentication services (e.g., Kerberos [NT94]) have addressed these obvious deficiencies.

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Available: http://www.ietf.org/rfc/rfc1094.txt
The dreaded specification; read it if you must, i.e., you are getting paid to read it. Hopefully, paid a lot. Cash money!

[V72] "La Begueule"
Francois-Marie Arouet a.k.a. Voltaire
Published in 1772
Voltaire said a number of clever things, this being but one example. For example, Voltaire also said "If you have two religions in your land, the two will cut each others throats; but if you have thirty religions, they will dwell in peace." What do you say to that, Democrats and Republicans?

49

The Andrew File System (AFS)

The Andrew File System was introduced at Carnegie-Mellon University (CMU) [1] in the 1980's [H+88]. Led by the well-known Professor M. Satyanarayanan of Carnegie-Mellon University ("Satya" for short), the main goal of this project was simple: **scale**. Specifically, how can one design a distributed file system such that a server can support as many clients as possible?

Interestingly, there are numerous aspects of design and implementation that affect scalability. Most important is the design of the **protocol** between clients and servers. In NFS, for example, the protocol forces clients to check with the server periodically to determine if cached contents have changed; because each check uses server resources (including CPU and network bandwidth), frequent checks like this will limit the number of clients a server can respond to and thus limit scalability.

AFS also differs from NFS in that from the beginning, reasonable user-visible behavior was a first-class concern. In NFS, cache consistency is hard to describe because it depends directly on low-level implementation details, including client-side cache timeout intervals. In AFS, cache consistency is simple and readily understood: when the file is opened, a client will generally receive the latest consistent copy from the server.

49.1 AFS Version 1

We will discuss two versions of AFS [H+88, S+85]. The first version (which we will call AFSv1, but actually the original system was called the ITC distributed file system [S+85]) had some of the basic design in place, but didn't scale as desired, which led to a re-design and the final protocol (which we will call AFSv2, or just AFS) [H+88]. We now discuss the first version.

[1] Though originally referred to as "Carnegie-Mellon University", CMU later dropped the hyphen, and thus was born the modern form, "Carnegie Mellon University." As AFS derived from work in the early 80's, we refer to CMU in its original fully-hyphenated form. See `https://www.quora.com/When-did-Carnegie-Mellon-University-remove-the-hyphen-in-the-university-name` for more details, if you are into really boring minutae.

```
TestAuth      Test whether a file has changed
              (used to validate cached entries)
GetFileStat   Get the stat info for a file
Fetch         Fetch the contents of file
Store         Store this file on the server
SetFileStat   Set the stat info for a file
ListDir       List the contents of a directory
```

Figure 49.1: **AFSv1 Protocol Highlights**

One of the basic tenets of all versions of AFS is **whole-file caching** on the **local disk** of the client machine that is accessing a file. When you `open()` a file, the entire file (if it exists) is fetched from the server and stored in a file on your local disk. Subsequent application `read()` and `write()` operations are redirected to the local file system where the file is stored; thus, these operations require no network communication and are fast. Finally, upon `close()`, the file (if it has been modified) is flushed back to the server. Note the obvious contrasts with NFS, which caches *blocks* (not whole files, although NFS could of course cache every block of an entire file) and does so in client *memory* (not local disk).

Let's get into the details a bit more. When a client application first calls `open()`, the AFS client-side code (which the AFS designers call **Venus**) would send a Fetch protocol message to the server. The Fetch protocol message would pass the entire pathname of the desired file (for example, `/home/remzi/notes.txt`) to the file server (the group of which they called **Vice**), which would then traverse the pathname, find the desired file, and ship the entire file back to the client. The client-side code would then cache the file on the local disk of the client (by writing it to local disk). As we said above, subsequent `read()` and `write()` system calls are strictly *local* in AFS (no communication with the server occurs); they are just redirected to the local copy of the file. Because the `read()` and `write()` calls act just like calls to a local file system, once a block is accessed, it also may be cached in client memory. Thus, AFS also uses client memory to cache copies of blocks that it has in its local disk. Finally, when finished, the AFS client checks if the file has been modified (i.e., that it has been opened for writing); if so, it flushes the new version back to the server with a Store protocol message, sending the entire file and pathname to the server for permanent storage.

The next time the file is accessed, AFSv1 does so much more efficiently. Specifically, the client-side code first contacts the server (using the TestAuth protocol message) in order to determine whether the file has changed. If not, the client would use the locally-cached copy, thus improving performance by avoiding a network transfer. The figure above shows some of the protocol messages in AFSv1. Note that this early version of the protocol only cached file contents; directories, for example, were only kept at the server.

TIP: MEASURE THEN BUILD (PATTERSON'S LAW)
One of our advisors, David Patterson (of RISC and RAID fame), used to always encourage us to measure a system and demonstrate a problem *before* building a new system to fix said problem. By using experimental evidence, rather than gut instinct, you can turn the process of system building into a more scientific endeavor. Doing so also has the fringe benefit of making you think about how exactly to measure the system before your improved version is developed. When you do finally get around to building the new system, two things are better as a result: first, you have evidence that shows you are solving a real problem; second, you now have a way to measure your new system in place, to show that it actually improves upon the state of the art. And thus we call this **Patterson's Law**.

49.2 Problems with Version 1

A few key problems with this first version of AFS motivated the designers to rethink their file system. To study the problems in detail, the designers of AFS spent a great deal of time measuring their existing prototype to find what was wrong. Such experimentation is a good thing, because **measurement** is the key to understanding how systems work and how to improve them; obtaining concrete, good data is thus a necessary part of systems construction. In their study, the authors found two main problems with AFSv1:

- **Path-traversal costs are too high**: When performing a Fetch or Store protocol request, the client passes the entire pathname (e.g., `/home/remzi/notes.txt`) to the server. The server, in order to access the file, must perform a full pathname traversal, first looking in the root directory to find `home`, then in `home` to find `remzi`, and so forth, all the way down the path until finally the desired file is located. With many clients accessing the server at once, the designers of AFS found that the server was spending much of its CPU time simply walking down directory paths.
- **The client issues too many TestAuth protocol messages**: Much like NFS and its overabundance of GETATTR protocol messages, AFSv1 generated a large amount of traffic to check whether a local file (or its stat information) was valid with the TestAuth protocol message. Thus, servers spent much of their time telling clients whether it was OK to used their cached copies of a file. Most of the time, the answer was that the file had not changed.

There were actually two other problems with AFSv1: load was not balanced across servers, and the server used a single distinct process per client thus inducing context switching and other overheads. The load

imbalance problem was solved by introducing **volumes**, which an administrator could move across servers to balance load; the context-switch problem was solved in AFSv2 by building the server with threads instead of processes. However, for the sake of space, we focus here on the main two protocol problems above that limited the scale of the system.

49.3 Improving the Protocol

The two problems above limited the scalability of AFS; the server CPU became the bottleneck of the system, and each server could only service 20 clients without becoming overloaded. Servers were receiving too many TestAuth messages, and when they received Fetch or Store messages, were spending too much time traversing the directory hierarchy. Thus, the AFS designers were faced with a problem:

THE CRUX: HOW TO DESIGN A SCALABLE FILE PROTOCOL
How should one redesign the protocol to minimize the number of server interactions, i.e., how could they reduce the number of TestAuth messages? Further, how could they design the protocol to make these server interactions efficient? By attacking both of these issues, a new protocol would result in a much more scalable version AFS.

49.4 AFS Version 2

AFSv2 introduced the notion of a **callback** to reduce the number of client/server interactions. A callback is simply a promise from the server to the client that the server will inform the client when a file that the client is caching has been modified. By adding this **state** to the system, the client no longer needs to contact the server to find out if a cached file is still valid. Rather, it assumes that the file is valid until the server tells it otherwise; insert analogy to **polling** versus **interrupts** here.

AFSv2 also introduced the notion of a **file identifier** (**FID**) (similar to the NFS **file handle**) instead of pathnames to specify which file a client was interested in. An FID in AFS consists of a volume identifier, a file identifier, and a "uniquifier" (to enable reuse of the volume and file IDs when a file is deleted). Thus, instead of sending whole pathnames to the server and letting the server walk the pathname to find the desired file, the client would walk the pathname, one piece at a time, caching the results and thus hopefully reducing the load on the server.

For example, if a client accessed the file `/home/remzi/notes.txt`, and `home` was the AFS directory mounted onto `/` (i.e., `/` was the local root directory, but `home` and its children were in AFS), the client would first Fetch the directory contents of `home`, put them in the local-disk cache, and set up a callback on `home`. Then, the client would Fetch the directory

Client (C_1)	Server
fd = open("/home/remzi/notes.txt", ...);	
Send Fetch (home FID, "remzi")	
	Receive Fetch request look for remzi in home dir establish callback(C_1) on remzi return remzi's content and FID
Receive Fetch reply write remzi to local disk cache record callback status of remzi Send Fetch (remzi FID, "notes.txt")	
	Receive Fetch request look for notes.txt in remzi dir establish callback(C_1) on notes.txt return notes.txt's content and FID
Receive Fetch reply write notes.txt to local disk cache record callback status of notes.txt local `open()` of cached notes.txt return file descriptor to application	
read(fd, buffer, MAX); perform local `read()` on cached copy	
close(fd); do local `close()` on cached copy if file has changed, flush to server	

fd = open("/home/remzi/notes.txt", ...);
Foreach dir (home, remzi)
if (callback(dir) == VALID)
use local copy for lookup(dir)
else
Fetch (as above)
if (callback(notes.txt) == VALID)
open local cached copy
return file descriptor to it
else
Fetch (as above) then open and return fd	

Figure 49.2: **Reading A File: Client-side And File Server Actions**

`remzi`, put it in the local-disk cache, and set up a callback on `remzi`. Finally, the client would Fetch `notes.txt`, cache this regular file in the local disk, set up a callback, and finally return a file descriptor to the calling application. See Figure 49.2 for a summary.

The key difference, however, from NFS, is that with each fetch of a directory or file, the AFS client would establish a callback with the server, thus ensuring that the server would notify the client of a change in its cached state. The benefit is obvious: although the *first* access to `/home/`

ASIDE: **CACHE CONSISTENCY IS NOT A PANACEA**
When discussing distributed file systems, much is made of the cache consistency the file systems provide. However, this baseline consistency does not solve all problems with regards to file access from multiple clients. For example, if you are building a code repository, with multiple clients performing check-ins and check-outs of code, you can't simply rely on the underlying file system to do all of the work for you; rather, you have to use explicit **file-level locking** in order to ensure that the "right" thing happens when such concurrent accesses take place. Indeed, any application that truly cares about concurrent updates will add extra machinery to handle conflicts. The baseline consistency described in this chapter and the previous one are useful primarily for casual usage, i.e., when a user logs into a different client, they expect some reasonable version of their files to show up there. Expecting more from these protocols is setting yourself up for failure, disappointment, and tear-filled frustration.

`remzi/notes.txt` generates many client-server messages (as described above), it also establishes callbacks for all the directories as well as the file notes.txt, and thus subsequent accesses are entirely local and require no server interaction at all. Thus, in the common case where a file is cached at the client, AFS behaves nearly identically to a local disk-based file system. If one accesses a file more than once, the second access should be just as fast as accessing a file locally.

49.5 Cache Consistency

When we discussed NFS, there were two aspects of cache consistency we considered: **update visibility** and **cache staleness**. With update visibility, the question is: when will the server be updated with a new version of a file? With cache staleness, the question is: once the server has a new version, how long before clients see the new version instead of an older cached copy?

Because of callbacks and whole-file caching, the cache consistency provided by AFS is easy to describe and understand. There are two important cases to consider: consistency between processes on *different* machines, and consistency between processes on the *same* machine.

Between different machines, AFS makes updates visible at the server and invalidates cached copies at the exact same time, which is when the updated file is closed. A client opens a file, and then writes to it (perhaps repeatedly). When it is finally closed, the new file is flushed to the server (and thus visibile). At this point, the server then "breaks" callbacks for any clients with cached copies; the break is accomplished by contacting each client and informing it that the callback it has on the file is no longer valid. This step ensures that clients will no longer read stale copies of the file; subsequent opens on those clients will require a re-fetch of the

$\mathbf{Client_1}$			$\mathbf{Client_2}$		Server	*Comments*
$\mathbf{P_1}$	$\mathbf{P_2}$	**Cache**	$\mathbf{P_3}$	**Cache**	**Disk**	
open(F)		-		-	-	File created
write(A)		A		-	-	
close()		A		-	A	
	open(F)	A		-	A	
	read() → A	A		-	A	
	close()	A		-	A	
open(F)		A		-	A	
write(B)		B		-	A	
	open(F)	B		-	A	Local processes
	read() → B	B		-	A	see writes immediately
	close()	B		-	A	
		B	open(F)	A	A	Remote processes
		B	read() → A	A	A	do not see writes...
		B	close()	A	A	
close()		B		~~A~~	B	... until close()
		B	open(F)	B	B	has taken place
		B	read() → B	B	B	
		B	close()	B	B	
		B	open(F)	B	B	
open(F)		B		B	B	
write(D)		D		B	B	
		D	write(C)	C	B	
		D	close()	C	C	
close()		D		~~C~~	D	
		D	open(F)	D	D	Unfortunately for P_3
		D	read() → D	D	D	the last writer wins
		D	close()	D	D	

Figure 49.3: **Cache Consistency Timeline**

new version of the file from the server (and will also serve to reestablish a callback on the new version of the file).

AFS makes an exception to this simple model between processes on the same machine. In this case, writes to a file are immediately visible to other local processes (i.e., a process does not have to wait until a file is closed to see its latest updates). This makes using a single machine behave exactly as you would expect, as this behavior is based upon typical UNIX semantics. Only when switching to a different machine would you be able to detect the more general AFS consistency mechanism.

There is one interesting cross-machine case that is worthy of further discussion. Specifically, in the rare case that processes on different machines are modifying a file at the same time, AFS naturally employs what is known as a **last writer wins** approach (which perhaps should be called **last closer wins**). Specifically, whichever client calls `close()` last will update the entire file on the server last and thus will be the "winning" file, i.e., the file that remains on the server for others to see. The result is a file that was generated in its entirety either by one client or the other. Note the difference from a block-based protocol like NFS: in NFS, writes of individual blocks may be flushed out to the server as each client is updating the file, and thus the final file on the server could end up as a mix of updates from both clients. In many cases, such a mixed file output

would not make much sense, i.e., imagine a JPEG image getting modified by two clients in pieces; the resulting mix of writes would not likely constitute a valid JPEG.

A timeline showing a few of these different scenarios can be seen in Figure 49.3. The columns show the behavior of two processes (P_1 and P_2) on $Client_1$ and its cache state, one process (P_3) on $Client_2$ and its cache state, and the server (Server), all operating on a single file called, imaginatively, `F`. For the server, the figure simply shows the contents of the file after the operation on the left has completed. Read through it and see if you can understand why each read returns the results that it does. A commentary field on the right will help you if you get stuck.

49.6 Crash Recovery

From the description above, you might sense that crash recovery is more involved than with NFS. You would be right. For example, imagine there is a short period of time where a server (S) is not able to contact a client (C1), for example, while the client C1 is rebooting. While C1 is not available, S may have tried to send it one or more callback recall messages; for example, imagine C1 had file F cached on its local disk, and then C2 (another client) updated F, thus causing S to send messages to all clients caching the file to remove it from their local caches. Because C1 may miss those critical messages when it is rebooting, upon rejoining the system, C1 should treat all of its cache contents as suspect. Thus, upon the next access to file F, C1 should first ask the server (with a TestAuth protocol message) whether its cached copy of file F is still valid; if so, C1 can use it; if not, C1 should fetch the newer version from the server.

Server recovery after a crash is also more complicated. The problem that arises is that callbacks are kept in memory; thus, when a server reboots, it has no idea which client machine has which files. Thus, upon server restart, each client of the server must realize that the server has crashed and treat all of their cache contents as suspect, and (as above) reestablish the validity of a file before using it. Thus, a server crash is a big event, as one must ensure that each client is aware of the crash in a timely manner, or risk a client accessing a stale file. There are many ways to implement such recovery; for example, by having the server send a message (saying "don't trust your cache contents!") to each client when it is up and running again, or by having clients check that the server is alive periodically (with a **heartbeat** message, as it is called). As you can see, there is a cost to building a more scalable and sensible caching model; with NFS, clients hardly noticed a server crash.

49.7 Scale And Performance Of AFSv2

With the new protocol in place, AFSv2 was measured and found to be much more scalable that the original version. Indeed, each server could

Workload	NFS	AFS	AFS/NFS
1. Small file, sequential read	$N_s \cdot L_{net}$	$N_s \cdot L_{net}$	1
2. Small file, sequential re-read	$N_s \cdot L_{mem}$	$N_s \cdot L_{mem}$	1
3. Medium file, sequential read	$N_m \cdot L_{net}$	$N_m \cdot L_{net}$	1
4. Medium file, sequential re-read	$N_m \cdot L_{mem}$	$N_m \cdot L_{mem}$	1
5. Large file, sequential read	$N_L \cdot L_{net}$	$N_L \cdot L_{net}$	1
6. Large file, sequential re-read	$N_L \cdot L_{net}$	$N_L \cdot L_{disk}$	$\frac{L_{disk}}{L_{net}}$
7. Large file, single read	L_{net}	$N_L \cdot L_{net}$	N_L
8. Small file, sequential write	$N_s \cdot L_{net}$	$N_s \cdot L_{net}$	1
9. Large file, sequential write	$N_L \cdot L_{net}$	$N_L \cdot L_{net}$	1
10. Large file, sequential overwrite	$N_L \cdot L_{net}$	$2 \cdot N_L \cdot L_{net}$	2
11. Large file, single write	L_{net}	$2 \cdot N_L \cdot L_{net}$	$2 \cdot N_L$

Figure 49.4: **Comparison: AFS vs. NFS**

support about 50 clients (instead of just 20). A further benefit was that client-side performance often came quite close to local performance, because in the common case, all file accesses were local; file reads usually went to the local disk cache (and potentially, local memory). Only when a client created a new file or wrote to an existing one was there need to send a Store message to the server and thus update the file with new contents.

Let us also gain some perspective on AFS performance by comparing common file-system access scenarios with NFS. Figure 49.4 (page 9) shows the results of our qualitative comparison.

In the figure, we examine typical read and write patterns analytically, for files of different sizes. Small files have N_s blocks in them; medium files have N_m blocks; large files have N_L blocks. We assume that small and medium files fit into the memory of a client; large files fit on a local disk but not in client memory.

We also assume, for the sake of analysis, that an access across the network to the remote server for a file block takes L_{net} time units. Access to local memory takes L_{mem}, and access to local disk takes L_{disk}. The general assumption is that $L_{net} > L_{disk} > L_{mem}$.

Finally, we assume that the first access to a file does not hit in any caches. Subsequent file accesses (i.e., "re-reads") we assume will hit in caches, if the relevant cache has enough capacity to hold the file.

The columns of the figure show the time a particular operation (e.g., a small file sequential read) roughly takes on either NFS or AFS. The rightmost column displays the ratio of AFS to NFS.

We make the following observations. First, in many cases, the performance of each system is roughly equivalent. For example, when first reading a file (e.g., Workloads 1, 3, 5), the time to fetch the file from the remote server dominates, and is similar on both systems. You might think AFS would be slower in this case, as it has to write the file to local disk; however, those writes are buffered by the local (client-side) file system cache and thus said costs are likely hidden. Similarly, you might think that AFS reads from the local cached copy would be slower, again be-

cause AFS stores the cached copy on disk. However, AFS again benefits here from local file system caching; reads on AFS would likely hit in the client-side memory cache, and performance would be similar to NFS.

Second, an interesting difference arises during a large-file sequential re-read (Workload 6). Because AFS has a large local disk cache, it will access the file from there when the file is accessed again. NFS, in contrast, only can cache blocks in client memory; as a result, if a large file (i.e., a file bigger than local memory) is re-read, the NFS client will have to re-fetch the entire file from the remote server. Thus, AFS is faster than NFS in this case by a factor of $\frac{L_{net}}{L_{disk}}$, assuming that remote access is indeed slower than local disk. We also note that NFS in this case increases server load, which has an impact on scale as well.

Third, we note that sequential writes (of new files) should perform similarly on both systems (Workloads 8, 9). AFS, in this case, will write the file to the local cached copy; when the file is closed, the AFS client will force the writes to the server, as per the protocol. NFS will buffer writes in client memory, perhaps forcing some blocks to the server due to client-side memory pressure, but definitely writing them to the server when the file is closed, to preserve NFS flush-on-close consistency. You might think AFS would be slower here, because it writes all data to local disk. However, realize that it is writing to a local file system; those writes are first committed to the page cache, and only later (in the background) to disk, and thus AFS reaps the benefits of the client-side OS memory caching infrastructure to improve performance.

Fourth, we note that AFS performs worse on a sequential file overwrite (Workload 10). Thus far, we have assumed that the workloads that write are also creating a new file; in this case, the file exists, and is then over-written. Overwrite can be a particularly bad case for AFS, because the client first fetches the old file in its entirety, only to subsequently overwrite it. NFS, in contrast, will simply overwrite blocks and thus avoid the initial (useless) read[2].

Finally, workloads that access a small subset of data within large files perform much better on NFS than AFS (Workloads 7, 11). In these cases, the AFS protocol fetches the entire file when the file is opened; unfortunately, only a small read or write is performed. Even worse, if the file is modified, the entire file is written back to the server, doubling the performance impact. NFS, as a block-based protocol, performs I/O that is proportional to the size of the read or write.

Overall, we see that NFS and AFS make different assumptions and not surprisingly realize different performance outcomes as a result. Whether these differences matter is, as always, a question of workload.

[2]We assume here that NFS reads are block-sized and block-aligned; if they were not, the NFS client would also have to read the block first. We also assume the file was *not* opened with the O_TRUNC flag; if it had been, the initial open in AFS would not fetch the soon to be truncated file's contents.

ASIDE: **THE IMPORTANCE OF WORKLOAD**

One challenge of evaluating any system is the choice of **workload**. Because computer systems are used in so many different ways, there are a large variety of workloads to choose from. How should the storage system designer decide which workloads are important, in order to make reasonable design decisions?

The designers of AFS, given their experience in measuring how file systems were used, made certain workload assumptions; in particular, they assumed that most files were not frequently shared, and accessed sequentially in their entirety. Given those assumptions, the AFS design makes perfect sense.

However, these assumptions are not always correct. For example, imagine an application that appends information, periodically, to a log. These little log writes, which add small amounts of data to an existing large file, are quite problematic for AFS. Many other difficult workloads exist as well, e.g., random updates in a transaction database.

One place to get some information about what types of workloads are common are through various research studies that have been performed. See any of these studies for good examples of workload analysis [B+91, H+11, R+00, V99], including the AFS retrospective [H+88].

49.8 AFS: Other Improvements

Like we saw with the introduction of Berkeley FFS (which added symbolic links and a number of other features), the designers of AFS took the opportunity when building their system to add a number of features that made the system easier to use and manage. For example, AFS provides a true global namespace to clients, thus ensuring that all files were named the same way on all client machines. NFS, in contrast, allows each client to mount NFS servers in any way that they please, and thus only by convention (and great administrative effort) would files be named similarly across clients.

AFS also takes security seriously, and incorporates mechanisms to authenticate users and ensure that a set of files could be kept private if a user so desired. NFS, in contrast, had quite primitive support for security for many years.

AFS also includes facilities for flexible user-managed access control. Thus, when using AFS, a user has a great deal of control over who exactly can access which files. NFS, like most UNIX file systems, has much less support for this type of sharing.

Finally, as mentioned before, AFS adds tools to enable simpler management of servers for the administrators of the system. In thinking about system management, AFS was light years ahead of the field.

49.9 Summary

AFS shows us how distributed file systems can be built quite differently than what we saw with NFS. The protocol design of AFS is particularly important; by minimizing server interactions (through whole-file caching and callbacks), each server can support many clients and thus reduce the number of servers needed to manage a particular site. Many other features, including the single namespace, security, and access-control lists, make AFS quite nice to use. The consistency model provided by AFS is simple to understand and reason about, and does not lead to the occasional weird behavior as one sometimes observes in NFS.

Perhaps unfortunately, AFS is likely on the decline. Because NFS became an open standard, many different vendors supported it, and, along with CIFS (the Windows-based distributed file system protocol), NFS dominates the marketplace. Although one still sees AFS installations from time to time (such as in various educational institutions, including Wisconsin), the only lasting influence will likely be from the ideas of AFS rather than the actual system itself. Indeed, NFSv4 now adds server state (e.g., an "open" protocol message), and thus bears an increasing similarity to the basic AFS protocol.

References

[B+91] "Measurements of a Distributed File System"
Mary Baker, John Hartman, Martin Kupfer, Ken Shirriff, John Ousterhout
SOSP '91, Pacific Grove, California, October 1991
An early paper measuring how people use distributed file systems. Matches much of the intuition found in AFS.

[H+11] "A File is Not a File: Understanding the I/O Behavior of Apple Desktop Applications"
Tyler Harter, Chris Dragga, Michael Vaughn,
Andrea C. Arpaci-Dusseau, Remzi H. Arpaci-Dusseau
SOSP '11, New York, New York, October 2011
Our own paper studying the behavior of Apple Desktop workloads; turns out they are a bit different than many of the server-based workloads the systems research community usually focuses upon. Also a good recent reference which points to a lot of related work.

[H+88] "Scale and Performance in a Distributed File System"
John H. Howard, Michael L. Kazar, Sherri G. Menees, David A. Nichols, M. Satyanarayanan, Robert N. Sidebotham, Michael J. West
ACM Transactions on Computing Systems (ACM TOCS), page 51-81, Volume 6, Number 1, February 1988
The long journal version of the famous AFS system, still in use in a number of places throughout the world, and also probably the earliest clear thinking on how to build distributed file systems. A wonderful combination of the science of measurement and principled engineering.

[R+00] "A Comparison of File System Workloads"
Drew Roselli, Jacob R. Lorch, Thomas E. Anderson
USENIX '00, San Diego, California, June 2000
A more recent set of traces as compared to the Baker paper [B+91], with some interesting twists.

[S+85] "The ITC Distributed File System: Principles and Design"
M. Satyanarayanan, J.H. Howard, D.A. Nichols, R.N. Sidebotham, A. Spector, M.J. West
SOSP '85, Orcas Island, Washington, December 1985
The older paper about a distributed file system. Much of the basic design of AFS is in place in this older system, but not the improvements for scale. The name change to "Andrew" is an homage to two people both named Andrew, Andrew Carnegie and Andrew Mellon. Thesee two rich dudes started the Carnegie Institute of Technology and the Mellon Institute of Industrial Research, respectively, which eventually merged to become what is now known as Carnegie Mellon University.

[V99] "File system usage in Windows NT 4.0"
Werner Vogels
SOSP '99, Kiawah Island Resort, South Carolina, December 1999
A cool study of Windows workloads, which are inherently different than many of the UNIX*-based studies that had previously been done.*

Homework

This section introduces `afs.py`, a simple AFS simulator you can use to shore up your knowledge of how the Andrew File System works. Read the README file for more details.

Questions

1. Run a few simple cases to make sure you can predict what values will be read by clients. Vary the random seed flag (`-s`) and see if you can trace through and predict both intermediate values as well as the final values stored in the files. Also vary the number of files (`-f`), the number of clients (`-C`), and the read ratio (`-r`, from between 0 to 1) to make it a bit more challenging. You might also want to generate slightly longer traces to make for more interesting interactions, e.g., (`-n 2` or higher).
2. Now do the same thing and see if you can predict each callback that the AFS server initiates. Try different random seeds, and make sure to use a high level of detailed feedback (e.g., `-d 3`) to see when callbacks occur when you have the program compute the answers for you (with `-c`). Can you guess exactly when each callback occurs? What is the precise condition for one to take place?
3. Similar to above, run with some different random seeds and see if you can predict the exact cache state at each step. Cache state can be observed by running with `-c` and `-d 7`.
4. Now let's construct some specific workloads. Run the simulation with `-A oa1:w1:c1,oa1:r1:c1` flag. What are different possible values observed by client 1 when it reads the file `a`, when running with the random scheduler? (try different random seeds to see different outcomes)? Of all the possible schedule interleavings of the two clients' operations, how many of them lead to client 1 reading the value 1, and how many reading the value 0?
5. Now let's construct some specific schedules. When running with the `-A oa1:w1:c1,oa1:r1:c1` flag, also run with the following schedules: `-S 01`, `-S 100011`, `-S 011100`, and others of which you can think. What value will client 1 read?
6. Now run with this workload: `-A oa1:w1:c1,oa1:w1:c1`, and vary the schedules as above. What happens when you run with `-S 011100`? What about when you run with `-S 010011`? What is important in determining the final value of the file?

50

Summary Dialogue on Distribution

Student: *Well, that was quick. Too quick, in my opinion!*

Professor: *Yes, distributed systems are complicated and cool and well worth your study; just not in this book (or course).*

Student: *That's too bad; I wanted to learn more! But I did learn a few things.*

Professor: *Like what?*

Student: *Well, everything can fail.*

Professor: *Good start.*

Student: *But by having lots of these things (whether disks, machines, or whatever), you can hide much of the failure that arises.*

Professor: *Keep going!*

Student: *Some basic techniques like retrying are really useful.*

Professor: *That's true.*

Student: *And you have to think carefully about protocols: the exact bits that are exchanged between machines. Protocols can affect everything, including how systems respond to failure and how scalable they are.*

Professor: *You really are getting better at this learning stuff.*

Student: *Thanks! And you're not a bad teacher yourself!*

Professor: *Well thank you very much too.*

Student: *So is this the end of the book?*

Professor: *I'm not sure. They don't tell me anything.*

Student: *Me neither. Let's get out of here.*

Professor: *OK.*

Student: *Go ahead.*

Professor: *No, after you.*

Student: *Please, professors first.*

Professor: *No, please, after you.*

Student: *(exasperated) Fine!*

Professor: *(waiting) ... so why haven't you left?*

Student: *I don't know how. Turns out, the only thing I can do is participate in these dialogues.*

Professor: *Me too. And now you've learned our final lesson...*