33

Event-based Concurrency (Advanced)

Thus far, we've written about concurrency as if the only way to build concurrent applications is to use threads. Like many things in life, this is not completely true. Specifically, a different style of concurrent programming is often used in both GUI-based applications [O96] as well as some types of internet servers [PDZ99]. This style, known as **event-based concurrency**, has become popular in some modern systems, including server-side frameworks such as **node.js** [N13], but its roots are found in C/UNIX systems that we'll discuss below.

The problem that event-based concurrency addresses is two-fold. The first is that managing concurrency correctly in multi-threaded applications can be challenging; as we've discussed, missing locks, deadlock, and other nasty problems can arise. The second is that in a multi-threaded application, the developer has little or no control over what is scheduled at a given moment in time; rather, the programmer simply creates threads and then hopes that the underlying OS schedules them in a reasonable manner across available CPUs. Given the difficulty of building a general-purpose scheduler that works well in all cases for all workloads, sometimes the OS will schedule work in a manner that is less than optimal. The crux:

THE CRUX:
HOW TO BUILD CONCURRENT SERVERS WITHOUT THREADS

How can we build a concurrent server without using threads, and thus retain control over concurrency as well as avoid some of the problems that seem to plague multi-threaded applications?

33.1 The Basic Idea: An Event Loop

The basic approach we'll use, as stated above, is called **event-based concurrency**. The approach is quite simple: you simply wait for something (i.e., an "event") to occur; when it does, you check what type of event it is and do the small amount of work it requires (which may include issuing I/O requests, or scheduling other events for future handling, etc.). That's it!

Before getting into the details, let's first examine what a canonical event-based server looks like. Such applications are based around a simple construct known as the **event loop**. Pseudocode for an event loop looks like this:

```
while (1) {
    events = getEvents();
    for (e in events)
        processEvent(e);
}
```

It's really that simple. The main loop simply waits for something to do (by calling `getEvents()` in the code above) and then, for each event returned, processes them, one at a time; the code that processes each event is known as an **event handler**. Importantly, when a handler processes an event, it is the only activity taking place in the system; thus, deciding which event to handle next is equivalent to scheduling. This explicit control over scheduling is one of the fundamental advantages of the event-based approach.

But this discussion leaves us with a bigger question: how exactly does an event-based server determine which events are taking place, in particular with regards to network and disk I/O? Specifically, how can an event server tell if a message has arrived for it?

33.2 An Important API: `select()` (or `poll()`)

With that basic event loop in mind, we next must address the question of how to receive events. In most systems, a basic API is available, via either the `select()` or `poll()` system calls.

What these interfaces enable a program to do is simple: check whether there is any incoming I/O that should be attended to. For example, imagine that a network application (such as a web server) wishes to check whether any network packets have arrived, in order to service them. These system calls let you do exactly that.

Take `select()` for example. The manual page (on Mac OS X) describes the API in this manner:

```
int select(int nfds,
           fd_set *restrict readfds,
           fd_set *restrict writefds,
           fd_set *restrict errorfds,
           struct timeval *restrict timeout);
```

The actual description from the man page: *select() examines the I/O descriptor sets whose addresses are passed in readfds, writefds, and errorfds to see if some of their descriptors are ready for reading, are ready for writing, or have*

ASIDE: **BLOCKING VS. NON-BLOCKING INTERFACES**
Blocking (or **synchronous**) interfaces do all of their work before returning to the caller; non-blocking (or **asynchronous**) interfaces begin some work but return immediately, thus letting whatever work that needs to be done get done in the background.

The usual culprit in blocking calls is I/O of some kind. For example, if a call must read from disk in order to complete, it might block, waiting for the I/O request that has been sent to the disk to return.

Non-blocking interfaces can be used in any style of programming (e.g., with threads), but are essential in the event-based approach, as a call that blocks will halt all progress.

an exceptional condition pending, respectively. The first nfds descriptors are checked in each set, i.e., the descriptors from 0 through nfds-1 in the descriptor sets are examined. On return, select() replaces the given descriptor sets with subsets consisting of those descriptors that are ready for the requested operation. select() returns the total number of ready descriptors in all the sets.

A couple of points about `select()`. First, note that it lets you check whether descriptors can be *read* from as well as *written* to; the former lets a server determine that a new packet has arrived and is in need of processing, whereas the latter lets the service know when it is OK to reply (i.e., the outbound queue is not full).

Second, note the timeout argument. One common usage here is to set the timeout to NULL, which causes `select()` to block indefinitely, until some descriptor is ready. However, more robust servers will usually specify some kind of timeout; one common technique is to set the timeout to zero, and thus use the call to `select()` to return immediately.

The `poll()` system call is quite similar. See its manual page, or Stevens and Rago [SR05], for details.

Either way, these basic primitives give us a way to build a non-blocking event loop, which simply checks for incoming packets, reads from sockets with messages upon them, and replies as needed.

33.3 Using `select()`

To make this more concrete, let's examine how to use `select()` to see which network descriptors have incoming messages upon them. Figure 33.1 shows a simple example.

This code is actually fairly simple to understand. After some initialization, the server enters an infinite loop. Inside the loop, it uses the `FD_ZERO()` macro to first clear the set of file descriptors, and then uses `FD_SET()` to include all of the file descriptors from `minFD` to `maxFD` in the set. This set of descriptors might represent, for example, all of the net-

```
#include <stdio.h>
#include <stdlib.h>
#include <sys/time.h>
#include <sys/types.h>
#include <unistd.h>

int main(void) {
    // open and set up a bunch of sockets (not shown)
    // main loop
    while (1) {
        // initialize the fd_set to all zero
        fd_set readFDs;
        FD_ZERO(&readFDs);

        // now set the bits for the descriptors
        // this server is interested in
        // (for simplicity, all of them from min to max)
        int fd;
        for (fd = minFD; fd < maxFD; fd++)
            FD_SET(fd, &readFDs);

        // do the select
        int rc = select(maxFD+1, &readFDs, NULL, NULL, NULL);

        // check which actually have data using FD_ISSET()
        int fd;
        for (fd = minFD; fd < maxFD; fd++)
            if (FD_ISSET(fd, &readFDs))
                processFD(fd);
    }
}
```

Figure 33.1: **Simple Code Using `select()`**

work sockets to which the server is paying attention. Finally, the server calls `select()` to see which of the connections have data available upon them. By then using `FD_ISSET()` in a loop, the event server can see which of the descriptors have data ready and process the incoming data.

Of course, a real server would be more complicated than this, and require logic to use when sending messages, issuing disk I/O, and many other details. For further information, see Stevens and Rago [SR05] for API information, or Pai et. al or Welsh et al. for a good overview of the general flow of event-based servers [PDZ99, WCB01].

33.4 Why Simpler? No Locks Needed

With a single CPU and an event-based application, the problems found in concurrent programs are no longer present. Specifically, because only one event is being handled at a time, there is no need to acquire or release locks; the event-based server cannot be interrupted by another thread because it is decidedly single threaded. Thus, concurrency bugs common in threaded programs do not manifest in the basic event-based approach.

TIP: DON'T BLOCK IN EVENT-BASED SERVERS
Event-based servers enable fine-grained control over scheduling of tasks. However, to maintain such control, no call that blocks the execution the caller can ever be made; failing to obey this design tip will result in a blocked event-based server, frustrated clients, and serious questions as to whether you ever read this part of the book.

33.5 A Problem: Blocking System Calls

Thus far, event-based programming sounds great, right? You program a simple loop, and handle events as they arise. You don't even need to think about locking! But there is an issue: what if an event requires that you issue a system call that might block?

For example, imagine a request comes from a client into a server to read a file from disk and return its contents to the requesting client (much like a simple HTTP request). To service such a request, some event handler will eventually have to issue an `open()` system call to open the file, followed by a series of `read()` calls to read the file. When the file is read into memory, the server will likely start sending the results to the client.

Both the `open()` and `read()` calls may issue I/O requests to the storage system (when the needed metadata or data is not in memory already), and thus may take a long time to service. With a thread-based server, this is no issue: while the thread issuing the I/O request suspends (waiting for the I/O to complete), other threads can run, thus enabling the server to make progress. Indeed, this natural **overlap** of I/O and other computation is what makes thread-based programming quite natural and straightforward.

With an event-based approach, however, there are no other threads to run: just the main event loop. And this implies that if an event handler issues a call that blocks, the *entire* server will do just that: block until the call completes. When the event loop blocks, the system sits idle, and thus is a huge potential waste of resources. We thus have a rule that must be obeyed in event-based systems: no blocking calls are allowed.

33.6 A Solution: Asynchronous I/O

To overcome this limit, many modern operating systems have introduced new ways to issue I/O requests to the disk system, referred to generically as **asynchronous I/O**. These interfaces enable an application to issue an I/O request and return control immediately to the caller, before the I/O has completed; additional interfaces enable an application to determine whether various I/Os have completed.

For example, let us examine the interface provided on Mac OS X (other systems have similar APIs). The APIs revolve around a basic structure, the `struct aiocb` or **AIO control block** in common terminology. A simplified version of the structure looks like this (see the manual pages for more information):

```
struct aiocb {
    int             aio_fildes;         /* File descriptor */
    off_t           aio_offset;         /* File offset */
    volatile void   *aio_buf;           /* Location of buffer */
    size_t          aio_nbytes;         /* Length of transfer */
};
```

To issue an asynchronous read to a file, an application should first fill in this structure with the relevant information: the file descriptor of the file to be read (`aio_fildes`), the offset within the file (`aio_offset`) as well as the length of the request (`aio_nbytes`), and finally the target memory location into which the results of the read should be copied (`aio_buf`).

After this structure is filled in, the application must issue the asynchronous call to read the file; on Mac OS X, this API is simply the **asynchronous read** API:

```
int aio_read(struct aiocb *aiocbp);
```

This call tries to issue the I/O; if successful, it simply returns right away and the application (i.e., the event-based server) can continue with its work.

There is one last piece of the puzzle we must solve, however. How can we tell when an I/O is complete, and thus that the buffer (pointed to by `aio_buf`) now has the requested data within it?

One last API is needed. On Mac OS X, it is referred to (somewhat confusingly) as `aio_error()`. The API looks like this:

```
int aio_error(const struct aiocb *aiocbp);
```

This system call checks whether the request referred to by `aiocbp` has completed. If it has, the routine returns success (indicated by a zero); if not, EINPROGRESS is returned. Thus, for every outstanding asynchronous I/O, an application can periodically **poll** the system via a call to `aio_error()` to determine whether said I/O has yet completed.

One thing you might have noticed is that it is painful to check whether an I/O has completed; if a program has tens or hundreds of I/Os issued at a given point in time, should it simply keep checking each of them repeatedly, or wait a little while first, or ... ?

To remedy this issue, some systems provide an approach based on the **interrupt**. This method uses UNIX **signals** to inform applications when an asynchronous I/O completes, thus removing the need to repeatedly ask the system. This polling vs. interrupts issue is seen in devices too, as you will see (or already have seen) in the chapter on I/O devices.

ASIDE: **UNIX SIGNALS**

A huge and fascinating infrastructure known as **signals** is present in all modern UNIX variants. At its simplest, signals provide a way to communicate with a process. Specifically, a signal can be delivered to an application; doing so stops the application from whatever it is doing to run a **signal handler**, i.e., some code in the application to handle that signal. When finished, the process just resumes its previous behavior.

Each signal has a name, such as **HUP** (hang up), **INT** (interrupt), **SEGV** (segmentation violation), etc; see the manual page for details. Interestingly, sometimes it is the kernel itself that does the signaling. For example, when your program encounters a segmentation violation, the OS sends it a **SIGSEGV** (prepending **SIG** to signal names is common); if your program is configured to catch that signal, you can actually run some code in response to this erroneous program behavior (which can be useful for debugging). When a signal is sent to a process not configured to handle that signal, some default behavior is enacted; for SEGV, the process is killed.

Here is a simple program that goes into an infinite loop, but has first set up a signal handler to catch SIGHUP:

```
#include <stdio.h>
#include <signal.h>

void handle(int arg) {
    printf("stop wakin’ me up...\n");
}

int main(int argc, char *argv[]) {
    signal(SIGHUP, handle);
    while (1)
        ; // doin’ nothin’ except catchin’ some sigs
    return 0;
}
```

You can send signals to it with the **kill** command line tool (yes, this is an odd and aggressive name). Doing so will interrupt the main while loop in the program and run the handler code `handle()`:

```
prompt> ./main &
[3] 36705
prompt> kill -HUP 36705
stop wakin’ me up...
prompt> kill -HUP 36705
stop wakin’ me up...
prompt> kill -HUP 36705
stop wakin’ me up...
```

There is a lot more to learn about signals, so much that a single page, much less a single chapter, does not nearly suffice. As always, there is one great source: Stevens and Rago [SR05]. Read more if interested.

In systems without asynchronous I/O, the pure event-based approach cannot be implemented. However, clever researchers have derived methods that work fairly well in their place. For example, Pai et al. [PDZ99] describe a hybrid approach in which events are used to process network packets, and a thread pool is used to manage outstanding I/Os. Read their paper for details.

33.7 Another Problem: State Management

Another issue with the event-based approach is that such code is generally more complicated to write than traditional thread-based code. The reason is as follows: when an event handler issues an asynchronous I/O, it must package up some program state for the next event handler to use when the I/O finally completes; this additional work is not needed in thread-based programs, as the state the program needs is on the stack of the thread. Adya et al. call this work **manual stack management**, and it is fundamental to event-based programming [A+02].

To make this point more concrete, let's look at a simple example in which a thread-based server needs to read from a file descriptor (`fd`) and, once complete, write the data that it read from the file to a network socket descriptor (`sd`). The code (ignoring error checking) looks like this:

```
int rc = read(fd, buffer, size);
rc = write(sd, buffer, size);
```

As you can see, in a multi-threaded program, doing this kind of work is trivial; when the `read()` finally returns, the code immediately knows which socket to write to because that information is on the stack of the thread (in the variable `sd`).

In an event-based system, life is not so easy. To perform the same task, we'd first issue the read asynchronously, using the AIO calls described above. Let's say we then periodically check for completion of the read using the `aio_error()` call; when that call informs us that the read is complete, how does the event-based server know what to do?

The solution, as described by Adya et al. [A+02], is to use an old programming language construct known as a **continuation** [FHK84]. Though it sounds complicated, the idea is rather simple: basically, record the needed information to finish processing this event in some data structure; when the event happens (i.e., when the disk I/O completes), look up the needed information and process the event.

In this specific case, the solution would be to record the socket descriptor (`sd`) in some kind of data structure (e.g., a hash table), indexed by the file descriptor (`fd`). When the disk I/O completes, the event handler would use the file descriptor to look up the continuation, which will return the value of the socket descriptor to the caller. At this point (finally), the server can then do the last bit of work to write the data to the socket.

33.8 What Is Still Difficult With Events

There are a few other difficulties with the event-based approach that we should mention. For example, when systems moved from a single CPU to multiple CPUs, some of the simplicity of the event-based approach disappeared. Specifically, in order to utilize more than one CPU, the event server has to run multiple event handlers in parallel; when doing so, the usual synchronization problems (e.g., critical sections) arise, and the usual solutions (e.g., locks) must be employed. Thus, on modern multicore systems, simple event handling without locks is no longer possible.

Another problem with the event-based approach is that it does not integrate well with certain kinds of systems activity, such as **paging**. For example, if an event-handler page faults, it will block, and thus the server will not make progress until the page fault completes. Even though the server has been structured to avoid *explicit* blocking, this type of *implicit* blocking due to page faults is hard to avoid and thus can lead to large performance problems when prevalent.

A third issue is that event-based code can be hard to manage over time, as the exact semantics of various routines changes [A+02]. For example, if a routine changes from non-blocking to blocking, the event handler that calls that routine must also change to accommodate its new nature, by ripping itself into two pieces. Because blocking is so disastrous for event-based servers, a programmer must always be on the lookout for such changes in the semantics of the APIs each event uses.

Finally, though asynchronous disk I/O is now possible on most platforms, it has taken a long time to get there [PDZ99], and it never quite integrates with asynchronous network I/O in as simple and uniform a manner as you might think. For example, while one would simply like to use the `select()` interface to manage all outstanding I/Os, usually some combination of `select()` for networking and the AIO calls for disk I/O are required.

33.9 Summary

We've presented a bare bones introduction to a different style of concurrency based on events. Event-based servers give control of scheduling to the application itself, but do so at some cost in complexity and difficulty of integration with other aspects of modern systems (e.g., paging). Because of these challenges, no single approach has emerged as best; thus, both threads and events are likely to persist as two different approaches to the same concurrency problem for many years to come. Read some research papers (e.g., [A+02, PDZ99, vB+03, WCB01]) or better yet, write some event-based code, to learn more.

References

[A+02] "Cooperative Task Management Without Manual Stack Management"
Atul Adya, Jon Howell, Marvin Theimer, William J. Bolosky, John R. Douceur
USENIX ATC '02, Monterey, CA, June 2002
This gem of a paper is the first to clearly articulate some of the difficulties of event-based concurrency, and suggests some simple solutions, as well explores the even crazier idea of combining the two types of concurrency management into a single application!

[FHK84] "Programming With Continuations"
Daniel P. Friedman, Christopher T. Haynes, Eugene E. Kohlbecker
In Program Transformation and Programming Environments, Springer Verlag, 1984
The classic reference to this old idea from the world of programming languages. Now increasingly popular in some modern languages.

[N13] "Node.js Documentation"
By the folks who build node.js
Available: http://nodejs.org/api/
One of the many cool new frameworks that help you readily build web services and applications. Every modern systems hacker should be proficient in frameworks such as this one (and likely, more than one). Spend the time and do some development in one of these worlds and become an expert.

[O96] "Why Threads Are A Bad Idea (for most purposes)"
John Ousterhout
Invited Talk at USENIX '96, San Diego, CA, January 1996
A great talk about how threads aren't a great match for GUI-based applications (but the ideas are more general). Ousterhout formed many of these opinions while he was developing Tcl/Tk, a cool scripting language and toolkit that made it 100x easier to develop GUI-based applications than the state of the art at the time. While the Tk GUI toolkit lives on (in Python for example), Tcl seems to be slowly dying (unfortunately).

[PDZ99] "Flash: An Efficient and Portable Web Server"
Vivek S. Pai, Peter Druschel, Willy Zwaenepoel
USENIX '99, Monterey, CA, June 1999
A pioneering paper on how to structure web servers in the then-burgeoning Internet era. Read it to understand the basics as well as to see the authors' ideas on how to build hybrids when support for asynchronous I/O is lacking.

[SR05] "Advanced Programming in the UNIX Environment"
W. Richard Stevens and Stephen A. Rago
Addison-Wesley, 2005
Once again, we refer to the classic must-have-on-your-bookshelf book of UNIX *systems programming. If there is some detail you need to know, it is in here.*

[vB+03] "Capriccio: Scalable Threads for Internet Services"
Rob von Behren, Jeremy Condit, Feng Zhou, George C. Necula, Eric Brewer
SOSP '03, Lake George, New York, October 2003
A paper about how to make threads work at extreme scale; a counter to all the event-based work ongoing at the time.

[WCB01] "SEDA: An Architecture for Well-Conditioned, Scalable Internet Services"
Matt Welsh, David Culler, and Eric Brewer
SOSP '01, Banff, Canada, October 2001
A nice twist on event-based serving that combines threads, queues, and event-based hanlding into one streamlined whole. Some of these ideas have found their way into the infrastructures of companies such as Google, Amazon, and elsewhere.

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...*

Device Drivers: Classes and Services

Introduction

Device drivers represent both:

generalizing abstractions

> gathering a myriad of very different devices together and synthesizing a few general classes (e.g. disks, network interfaces, graphics adaptors) and standard models, behaviors and interfaces to be implemented by all drivers for a given class.

simplifying abstractions

> providing an implemention of standard class interfaces while opaquely encapsulating the details of how to effectively and efficiently use a particular device.

For reasons of performance and control, Operating Systems tend not to be implemented in object oriented languages (does anybody remember JavaOS?). Yet despite being implemented in simpler langauges (often C), Operating Systems, in device drivers, offer highly evolved examples of a different realization of class interfaces, derivation, and inheritance.

Whether we are talking about storage, networking, video, or human-interface, the number of available devices is huge and growing. The number and diversity of these devices creates tremendous demands for object oriented code reuse:

- We want the system to behave similarly, no matter what the underlying devices were being used to provide storage, networking, etc. To ensure this, we would like most of the higher level functionality to be implemented in common, higher level modules.
- We would like to minimize the cost of developing drivers to support new devices. This is most easily done if the majority of the functionality is implemented in common code that can be inherrited by the individual drivers.
- As system functionality and performance are improved, we would like to ensure that those benefits accrue not only to new device drivers, but also to older device drivers.

These needs can be satisfied by implementing the higher level functionality (associated with each general class of device) in common code that uses per-device implementations of a standard sub-class driver to operate over a particular device. This requires:

- deriving device-driver sub-classes for each of the major classes of device.
- defining sub-class specific interfaces to be implemented by the drivers for all devices in each of those classes.
- creating per-device implementations of those standard sub-class interfaces.

Major Driver Classes

In the earliest versions of Unix, all devices were divided into two fundamental classes, which are still present in all Unix derivatives:

block devices

These are random-access devices, addressable in fixed size (e.g. 512 byte, 4K byte) blocks. Their drivers implement a *request* method to enqueue asynchronous DMA requests. The request descriptor included information about the desired operation (e.g. byte count, target device, disk address and in-memory buffer address) as well as completion information (how much data was transferred, error indications) and a condition variable the requestor could use to await the eventual completion of the request.

A read or write request could be issued for any number of blocks, but in most cases a large request would be broken into multiple single-block requests, each of which would be passed, block at a time, through the system buffer cache. For this reason block device drivers also implement a *fsync* method to flush out any buffered writes.

character devices

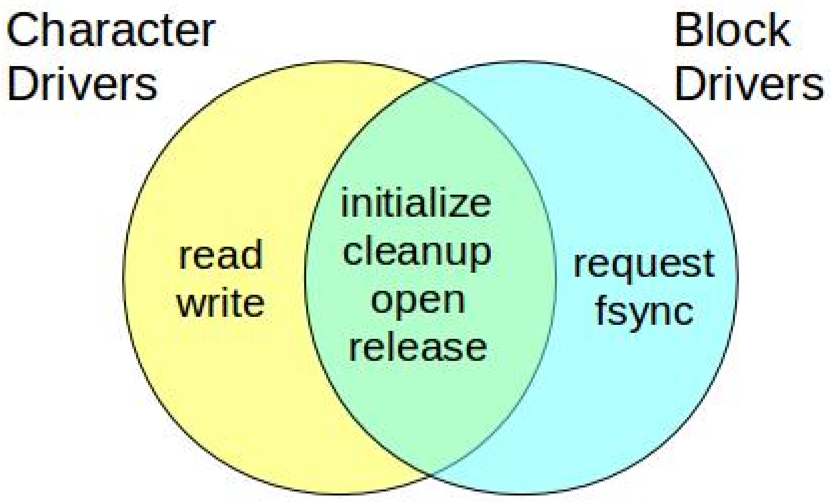
These devices may be sequential access, or may be byte-addressable. They support the standard synchronous *read(2)*, *write(2)* and (indirectly) *seek(2)* operations.

For devices that supported DMA, read and write operations were expected to be done as a single (potentially very large) DMA transfer between the device and the buffers in user address space.

A key point here is that, even in the oldest Unix systems, device drivers were divided into distinct classes (implementing different interfaces) based on the needs of distinct classes of clients:

- Block devices were designed to be used, within the operating system, by file systems, to access disks. Forcing all I/O to go through the system buffer cache is almost surely the right thing to do with file system I/O.
- Character devices were designed to be used directly by applications. The potential for large DMA transfers directly between the device and user-space buffers meant that character (or *raw*) I/O operations might be much more efficient than the corresponding requests to a block device.

These two major classes of device were not mutually exclusive. A single driver could export both block and character interfaces. A file system would be mounted on top of the block device, while back-up and integrity-checking software might access the disk through its (potentially much more efficient) character device*. All device drivers support *initialize* and *cleanup* methods (for dynamic module loading and unloading), *open* and *release* methods (roughly corresponding to the *open(2)* and *close(2)* system calls), and an optional catch-all *ioctl(2)* method.

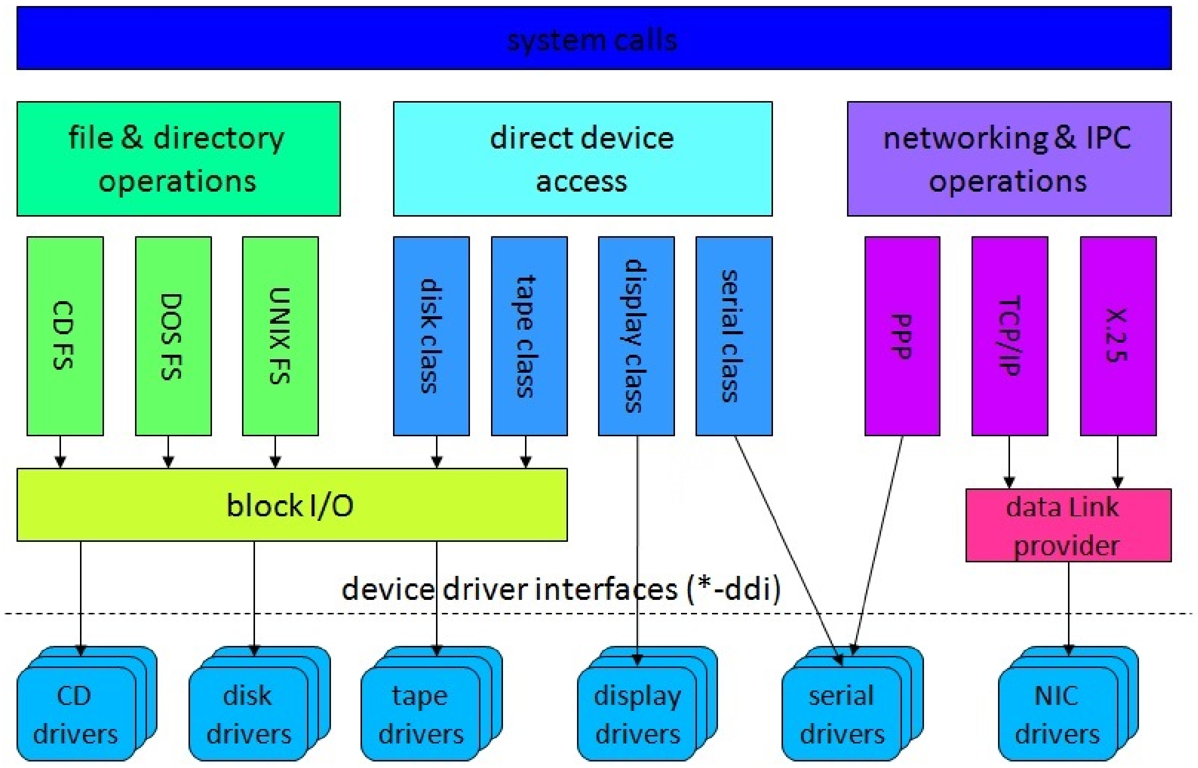


*In more contemporary systems, it is possible for a client to specify that block I/O should not be passed through the system buffer cache.

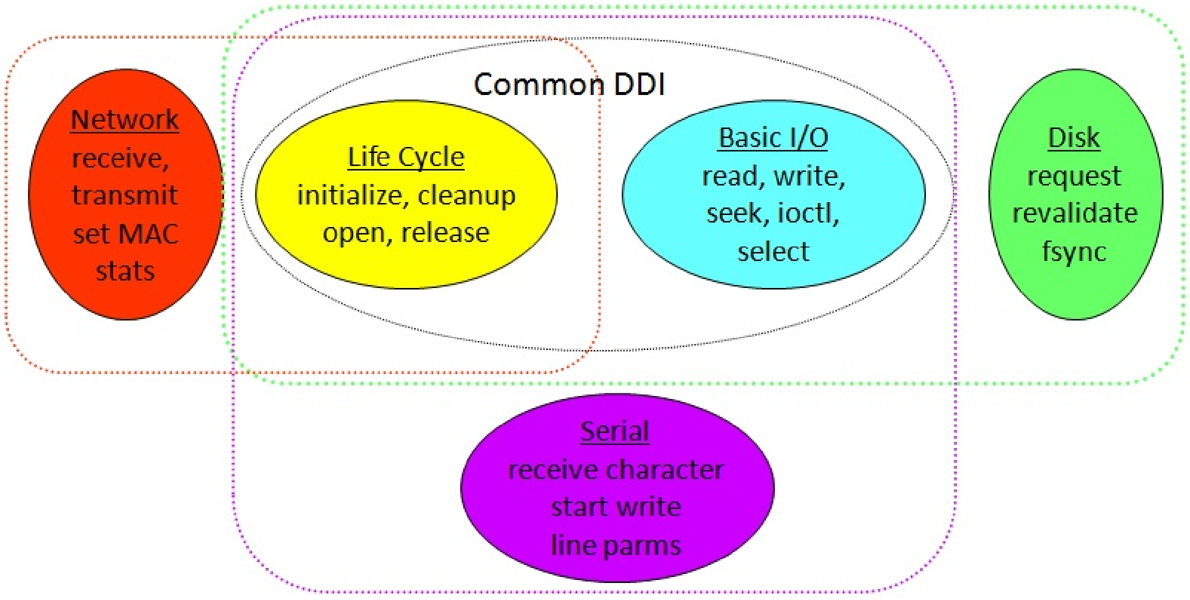
Driver sub-classes

Given the fundamental importance of file systems and disks to operating systems, it is not surprising that block device drivers would have been singled out as a special sub-class in even the earliest versions of Unix. But as system functionality evolved, the operating system began to implement higher level services for other sub-classes of devices:

- input editing and outut translation for terminals and virtual terminals
- address binding and packet sending/receipt for network interfaces
- display mapping and window management for graphics adaptors
- character-set mapping for keyboards
- cursor positioning for pointing devices
- sample mixing, volume and equalization for sound devices



And as each of these sub-systems evolved, new device driver methods* were defined to enable more effective communication between the higher level frameworks and the lower level device drivers in each sub-class. Each of these sub-class specific interfaces is referred to as a *Device Driver Interface* (DDI).



*It should be noted that in some cases, the higher frameworks have been implemented in user-mode, so that some of the new interfaces have been specified as behavior rather than new methods.

The rewards for this structure are:

- Sub-class drivers become easier to implement because so much of the important functionality is implemented in higher level software.
- The system behaves identically over a wide range of different devices.
- Most functionality enhancements to will be in the higher level code, and so should automatically work on all devices within that sub-class.

But the price for these rewards is that all device drivers must implement exactly the same interfaces:

- if a driver does not (correctly) implement the standard interfaces for its device sub-class, it will not work with the higher level software.
- if a driver implements additional functionality (not defined in the standard interfaces for its device sub-class), those features will not be exploited by the standard higher level software.

Services for Device Drivers

It is nearly impossible to implement a completely self-contained device driver. Most device drivers are likely to require a range of resources and services from the operating system:

- dynamic memory allocation
- I/O and bus resource allocation and management
- condition variable operations (wait and signal)
- mutual exclusion
- control of, and being called to service interrupts
- DMA target pin/release, scatter/gather map management
- configuration/registry services

The collection of services, exposed by the operating system for use by device drivers is sometimes referred to as the *Driver-Kernel Interface* (DKI). Interface stability for DKI functions is every bit as important as it is for the DDI entry points. If an operating system eliminates or incompatibly changes a DKI function, device drivers that depend on that function may cease working. The requirement to maintain stable DKI entry points may greatly constrain our ability to evolve our operating system implementation. Similar issues arrise for other classes of dynamically loadable kernel modules (such as network protocols and file systems).

Conclusion

Device drivers demonstrate an evolution from a basic super-class (character devices) into an ever-expanding hierarchy of derived sub-classes. But unlike traditional class derivation, where sub-class implementations inherit most of their implemntation from their parent, we see a different sort of inheritance. While each new sub-class and instance is likely to be a new implementation, what they inherit is pre-existing higher level frameworks that do do most of their work for them.

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-

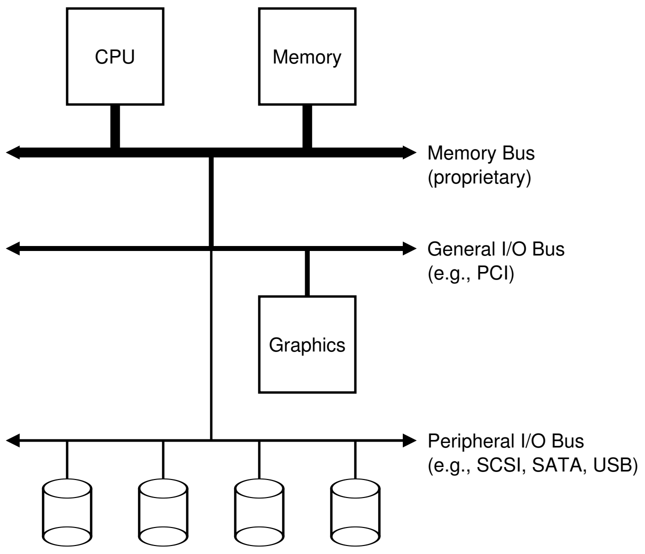


Figure 36.1: **Prototypical System Architecture**

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.

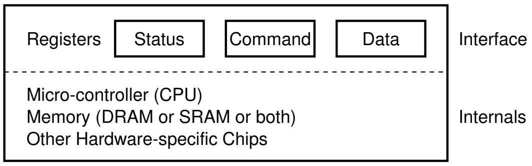


Figure 36.2: **A Canonical Device**

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.

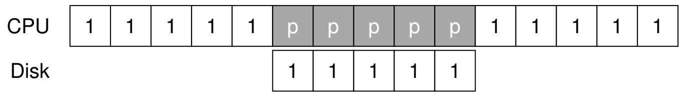
THE CRUX: HOW TO AVOID THE COSTS OF POLLING

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

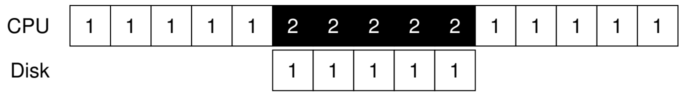
The invention that many engineers came upon years ago to improve this interaction is something we've seen already: the **interrupt**. Instead of polling the device repeatedly, the OS can issue a request, put the calling process to sleep, and context switch to another task. When the device is finally 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.

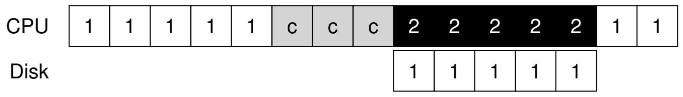
overlap, are best. If the speed of the device is not known, or sometimes fast and sometimes slow, it may be best to use a **hybrid** that polls for a little while and then, if the device is not yet finished, uses interrupts. This **two-phased** approach may achieve the best of both worlds.

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:



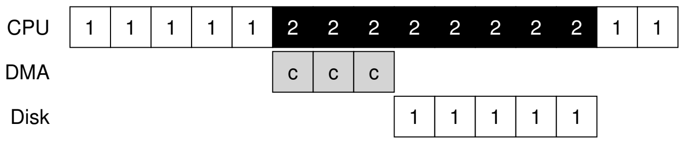
In the timeline, Process 1 is running and then wishes to write some data to the disk. It then initiates the I/O, which must copy the data from memory to the device explicitly, one word at a time (marked c in the diagram). When the copy is complete, the I/O begins on the disk and the CPU can finally be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

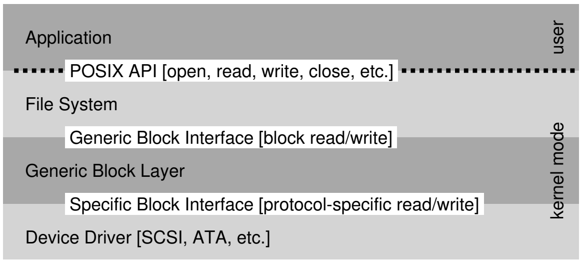

Application
user
POSIX API [open, read, write, close, etc.]
File System
Generic Block Interface [block read/write]
kernel mode
Generic Block Layer
Specific Block Interface [protocol-specific read/write]
Device Driver [SCSI, ATA, etc.]

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 not busy and READY.
- **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

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Irfan Ahmad, Ajay Gulati, Ali Mashtizadeh
USENIX '11
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**).

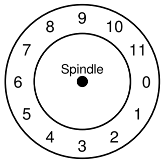

Figure 37.1: **A Disk With Just A Single Track**

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.

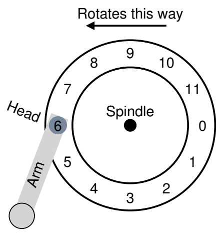

Figure 37.2: **A Single Track Plus A Head**

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).

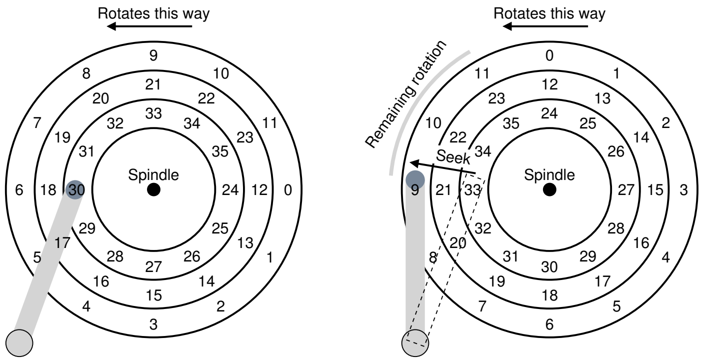

Figure 37.3: **Three Tracks Plus A Head (Right: With Seek)**

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.

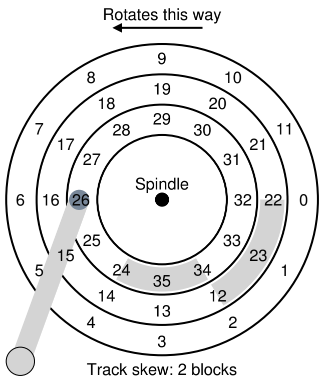

Figure 37.4: **Three Tracks: Track Skew Of 2**

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.

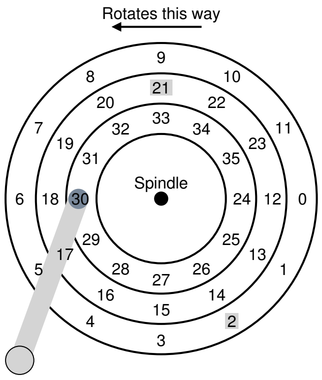

Figure 37.7: **SSTF: Scheduling Requests 21 And 2**

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?

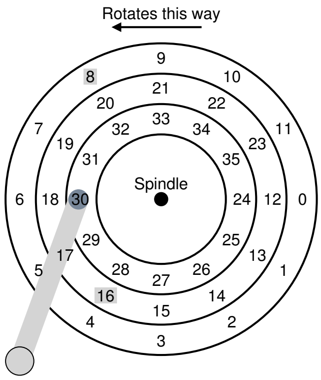

Figure 37.8: **SSTF: Sometimes Not Good Enough**

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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D. Jacobson, J. Wilkes
Technical Report HPL-CSP-91-7rev1, Hewlett-Packard (February 1991)
A more modern take on disk scheduling. It remains a technical report (and not a published paper) because the authors were scooped by Seltzer et al. [SCO90].

[RW92] "An Introduction to Disk Drive Modeling"
C. Ruemmler, J. Wilkes
IEEE Computer, 27:3, pp. 17-28, March 1994
A terrific introduction to the basics of disk operation. Some pieces are out of date, but most of the basics remain.

[SCO90] "Disk Scheduling Revisited"
Margo Seltzer, Peter Chen, John Ousterhout
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_cheetah_15k_5.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_barracuda_es.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?

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 challenging 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?

Dynamically Loadable Kernel Modules

Introduction

We often design systems to provide a common framework, but that expect problem-specific implementations to be provided at a later time. This approach is embraced by several standard design patterns (e.g. Strategy, Factory, Plug-In). These approaches have a few key elements in common:

- all implementations provide similar functionality in accordance with a common *interface*.
- the selection of a particular implementation can be deffered until run time.
- decoupling the overarching service from the plug-in implementations makes it possible for a system to work with a potentially open-ended set of of algorithms or adaptors.

In most programs the set of available implementations is locked in at build-time. But many systems have the ability to select and load new implementations at run time as *dynamically loadable modules*. We see this with browser plug-ins. Operating systems may also support many different types of dynamically loadable modules (e.g. file systems, network protocols, device drivers). Device drivers are a particularly important and rich class of dynamically loadable modules ... and therefore a good example to study.

There are several compelling reasons for wanting device drivers to be dynamically loadable:

- the number of possible I/O devices is far too large to build all of them in to the operating system.
- if we want an operating system to automatically work on any computer, it must have the ability to automatically identify and load the required device drivers.
- many devices (e.g. USB) are hot-pluggable, and we cannot know what drivers are required until the the associated device is plugged in to the computer.
- new devices become available long after the operating system has been shipped, and so must be *after market* addable.
- most device drivers are developed by the hardware manufacturers, and delivered to customers independently from the operating system.

Choosing Which Module to Load

In the abstract, a program needs an implementation and calls a *Factory* to obtain it. But how does the Factory know what implementation class to instantiate?

- for a browser plugin, there might be a MIME-type assocated with the data to be handled, and the browser can consult a registry to find the plug-in associated with that MIME-type. This is a very general mechanism ... but it presumes that data is tagged with a type, and that somebody is maintaining a tag-to-plugin registery.
- at the other extreme, the Factory could load all of the known plug-ins and call a *probe* method in each to see which (if any) of the plug-ins could identify the data and claim responsibility for handling it.

Long ago, dynamically loadable device drivers used the probing process, but this was both unreliable (might incorrectly accept the wrong device) and dangerious (touching random registers in random devices). Today most I/O busses support self-identifying devices. Each device has type, model, and even serial number information that can be queried in a standard way (e.g. by *walking the configuration space*). This information can be used, in combination with a device driver registry, to automatically select a driver for a given device. These registries may support precedence rules that can chose the best from among multiple competing drivers (e.g. a generic VGA driver, a GeForce driver, and a GeForce GTX 980 driver).

Loading a New Module

In many cases, the module to be loaded may be entirely self-contained (it makes no calls outside of itself) or uses only standard shared libraries (which are expected to be mapped in to the address space at well known locations). In these cases loading a new module is as simple as allocating some memory (or address space) and reading the new module into it.

In many cases (including device drivers and file systems) the dynamically loaded module may need to make use of other functions (e.g. memory allocation, synchronization, I/O) in the program into which it is being added. This means that the module to be loaded will (like an object module) have unresolved external references, and requires a *run-time loader* (a simplified linkage editor) that can look up and adjust all of those references as the new module is being loaded.

Note, however, that these references can only be from the dynamically loaded module into the program into which it is loaded (e.g. the operating system). The main program can never have any direct references into the dynamically loaded module ... because the dynamically loaded module may not always be there.

Initialization and Registration

If the operating system is not allowed to have any direct references into a dynamically loadable module, how can the new module be used? When the run-time loader is invoked to load a new dynamically loadable module, it is common for it to return a vector that contains a pointer to at least one method: an initialization function.

After the module has been loaded into memory, the main program calls its initialization method. For a dynamically loaded device driver, the initialization method might:

- allocate memory and initialize driver data structures.
- allocate I/O resources (e.g. bus addresses, interrupt levels, DMA channels) and assign them to the devices to be managed.
- register all of the device instances it supports. Part of this registration would involve providing a vector (of pointers to standard device drvier entry points) that client software could call in order to use these device instances.

Device instance configuration and initialization is another area where self-identifying devices have made it much easier to implement dynamically loaded device drivers:

- Long ago, devices were configured for particular bus addresses and interrupt levels with mechanical switches, that would be set before the card was plugged in to the bus. These resource assignments would be recorded in a system configuration table, which would be compiled (or read during system start-up) into the operating system, and used to select and configure corresponding device driver instances.
- More contemporary busses (like PCIe or USB) provide mechanisms to discover all of the available devices and learn what resources (e.g. bus addresses, DMA channels, interrupt levels) they require. The device driver can then allocate these resources from the associated bus driver, and assign the required resources to each device.

Using a Dynamically Loaded Module

The operating system will provide some means by which processes can open device instances. In Linux the OS exports a pseudo file system (/dev) that is full of *special files*, each one associated with a registered device instance. When a process attempts to open one of those special files, the operating system creates a reference from the open file instance to the registered device instance. From then on, when ever the process issues a *read(2)*, *write(2)*, or *ioctl(2)* system call on that file descriptor, the operating system forwards that call to the appropriate device driver entry point.

A similar approach is used when higher level frameworks (e.g. terminal sessions, network protocols or file systems) are implemented on top of a device. Each of those services maintains open references to the underlying devices, and when they need to talk to the device (e.g. to queue an I/O request or send a packet) the OS forwards that call to the appropriate device driver entry point.

The system often maintains a table of all registered device instances and the associated device driver entry points for each standard operation. When ever a request is to be made for any device, the operating system can simply index into this table by a device identifier to look up the address of the entry point to be called. Such mechanisms for registering heterogenous implementations and forwarding requests to the correct entry point are often referred to as *federation frameworks* because they combine a set of independent implementations into a single unified framework.

Unloading

When all open file descriptors into those devices have been closed and the driver is no-longer needed, the operating system can call an shut-down method that will cause the driver to:

- un-register itself as a device driver
- shut down the devices it had been managing
- return all allocated memory and I/O resources back to the operating system.

After which, the module can be safely unloaded and that memory freed as well.

The Criticality of Stable Interfaces

All of this his completely dependent on stable and well specified interfaces:

- the set of entry-points for any class of device driver must be well defined, and all drivers must compatably implement all of the relevant interfaces.
- the set of functions within the main program (OS) that the dynamically loaded modules are allowed to call must be well defined, and the interfaces to each of those functions must be stable.

If one device driver did not implement a standard entry point in the standard way, clients of that device would not work. Some functionality may be optional, and it may be acceptable for a device driver to refuse some requests. But this may make the application responsible for dealing with version some incompatabilities.

If an operating system does not implement some standard service function (e.g. memory allocation) in the standard way, a device driver written to that interface standard may not work when loaded into the non-compliant operating system.

There is often a tension between the conflicting needs to support new hardware and software features while retaining compatability with old device drivers.

Hot-Pluggable Devices and Drivers

One of the major advantages of dynamically loadable modules is that they can be loaded at any time; not merely during start-up. Hot-plug busses (e.g. USB) can generate events whenever a device is added to, or removed from the bus. In many systems a hot-plug manager:

- subscribes to hot-plug events.

- when a new device is inserted, walks the configuration space to identify the new device, finds, and loads the appropriate driver.
- when a device is removed, finds the associated driver and calls its *removal* method to inform it that the device is no longer on the bus.

Hot-plugable busses often have multiple power levels, and a newly inserted device may receive only enough power to enable it to be queried and configured. When the driver is ready to start using the device, it can instruct the bus to fully power the device. Some hot-pluggable busses also have mechanical safety interlocks to prevent a device from being removed while it is still in use. In these cases the driver must shut down and release the device before it can be removed.

Summary

This discussion has focused on dynamically loadable device drivers, but most of the issues (selecting the module to be loaded, dependencies of the loaded module on the host program, initializing and shutting down dynamically loaded modules, binding clients to dynamically loaded modules, and defining and managing the interfaces between the loaded and hosting modules) are applicable to a much wider range of dynamically loadable modules.

Stable and well standardized interfaces are critical to any such framework:

- the methods to be implemented by the dynamically loaded modules.
- the services that can be used by the dynamically loaded modules.