PART III
The Cloud Native Attributes

CHAPTER 6

It's All About Dependability

The most important property of a program is whether it accomplishes the intention of its user.[1]

—C.A.R. Hoare, *Communications of the ACM (October 1969)*

Professor Sir Charles Antony Richard (Tony) Hoare is a brilliant guy. He invented quicksort, authored Hoare Logic for reasoning about the correctness of computer programs, and created the formal language "communicating sequential processes" (CSP) that inspired Go's beloved concurrency model. Oh, and he developed the structured programming paradigm[2] that forms the foundation of all modern programming languages in common use today. He also invented the null reference. Please don't hold that against him, though. He publicly apologized[3] for it in 2009, calling it his "billion-dollar mistake."

Tony Hoare literally invented programming as we know it. So when he says that the single most important property of a program is whether it accomplishes the intention of its user, you can take that on some authority. Think about this for a second: Hoare specifically (and quite rightly) points out that it's the intention of a program's *users*—not its *creators*—that dictates whether a program is performing correctly. How inconvenient that the intentions of a program's users aren't always the same as those of its creator!

1 Hoare, C.A.R. "An Axiomatic Basis for Computer Programming.". *Communications of the ACM*, vol. 12, no. 10, October 1969, pp. 576–583. *https://oreil.ly/jOwO9*.

2 When Edsger W. Dijkstra coined the expression "GOTO considered harmful," he was referencing Hoare's work in structured programming.

3 Hoare, Tony. "Null References: The Billion Dollar Mistake." *InfoQ.com*. 25 August 2009. *https://oreil.ly/4QWS8*.

Given this assertion, it stands to reason that a user's first expectation about a program is that *the program works*. But when is a program "working"? This is actually a pretty big question, one that lies at the heart of cloud native design. The first goal of this chapter is to explore that very idea, and in the process, introduce concepts like "dependability" and "reliability" that we can use to better describe (and meet) user expectations. Finally, we'll briefly review a number of practices commonly used in cloud native development to ensure that services meet the expectations of its users. We'll discuss each of these in-depth throughout the remainder of this book.

What's the Point of Cloud Native?

In Chapter 1 we spent a few pages defining "cloud native," starting with the Cloud Native Computing Foundation's definition and working forward to the properties of an ideal cloud native service. We spent a few more pages talking about the pressures that have driven cloud native to be a thing in the first place.

What we didn't spend so much time on, however, was the *why* of cloud native. Why does the concept of cloud native even exist? Why would we even want our systems to be cloud native? What's its purpose? What makes it so special? Why should I care?

So, why *does* cloud native exist? The answer is actually pretty straightforward: it's all about dependability. In the first part of this chapter, we'll dig into the concept of dependability, what it is, why it's important, and how it underlies all the patterns and techniques that we call cloud native.

It's All About Dependability

Holly Cummins, the worldwide development community practice lead for the IBM Garage, famously said that "if cloud native has to be a synonym for anything, it would be idempotent."[4] Cummins is absolutely brilliant, and has said a lot of absolutely brilliant things,[5] but I think she only has half of the picture on this one. I think that idempotence is very important—perhaps even necessary for cloud native—but not sufficient. I'll elaborate.

The history of software, particularly the network-based kind, has been one of struggling to meet the expectations of increasingly sophisticated users. Long gone are the days when a service could go down at night "for maintenance." Users today rely heavily on the services they use, and they expect those services to be available and to respond promptly to their requests. Remember the last time you tried to start a Netflix movie and it took the longest five seconds of your life? Yeah, that.

4 *Cloud Native Is About Culture, Not Containers*. Cummins, Holly. Cloud Native London 2018.

5 If you ever have a chance to see her speak, I strongly recommend you take it.

Users don't care that your services have to be maintained. They won't wait patiently while you hunt down that mysterious source of latency. They just want to finish binge-watching the second season of Breaking Bad.[6]

All of the patterns and techniques that we associate with cloud native—*every single one*—exist to allow services to be deployed, operated, and maintained at scale in unreliable environments, driven by the need to produce dependable services that keep users happy.

In other words, I think that if "cloud native" has to be a synonym for anything, it would be "dependability."

What Is Dependability and Why Is It So Important?

I didn't choose the word "dependability" arbitrarily. It's actually a core concept in the field of *systems engineering*, which is full of some very smart people who say some very smart things about the design and management of complex systems. The concept of dependability in a computing context was first rigorously defined by Jean-Claude Laprie about 35 years ago,[7] who defined a system's dependability according to the expectations of its users. Laprie's original definition has been tweaked and extended over the years by various authors, but here's my favorite:

> The dependability of a computer system is its ability to avoid failures that are more frequent or more severe, and outage durations that are longer, than is acceptable to the user(s).[8]
>
> —Fundamental Concepts of Computer System Dependability (2001)

In other words, a dependable system consistently does what its users expect and can be quickly fixed when it doesn't.

By this definition, a system is dependable only when it can *justifiably* be trusted. Obviously, a system can't be considered dependable if it falls over any time one of its components glitch, or if it requires hours to recover from a failure. Even if it's been running for months without interruption, an undependable system may still be one bad day away from catastrophe: lucky isn't dependable.

Unfortunately, it's hard to objectively gauge "user expectations." For this reason, as illustrated in Figure 6-1, dependability is an umbrella concept encompassing several

6 Remember what Walt did to Jane that time? That was so messed up.

7 Laprie, J-C. "Dependable Computing and Fault Tolerance: Concepts and Terminology." *FTCS-15 The 15th Int'l Symposium on Fault-Tolerant Computing*, June 1985, pp. 2–11. *https://oreil.ly/UZFFY*.

8 A. Avižienis, J. Laprie, and B. Randell. "Fundamental Concepts of Computer System Dependability." *Research Report No. 1145, LAAS-CNRS*, April 2001. *https://oreil.ly/4YXd1*.

more specific and quantifiable attributes—availability, reliability, and maintainability—all of which are subject to similar threats that may be overcome by similar means.

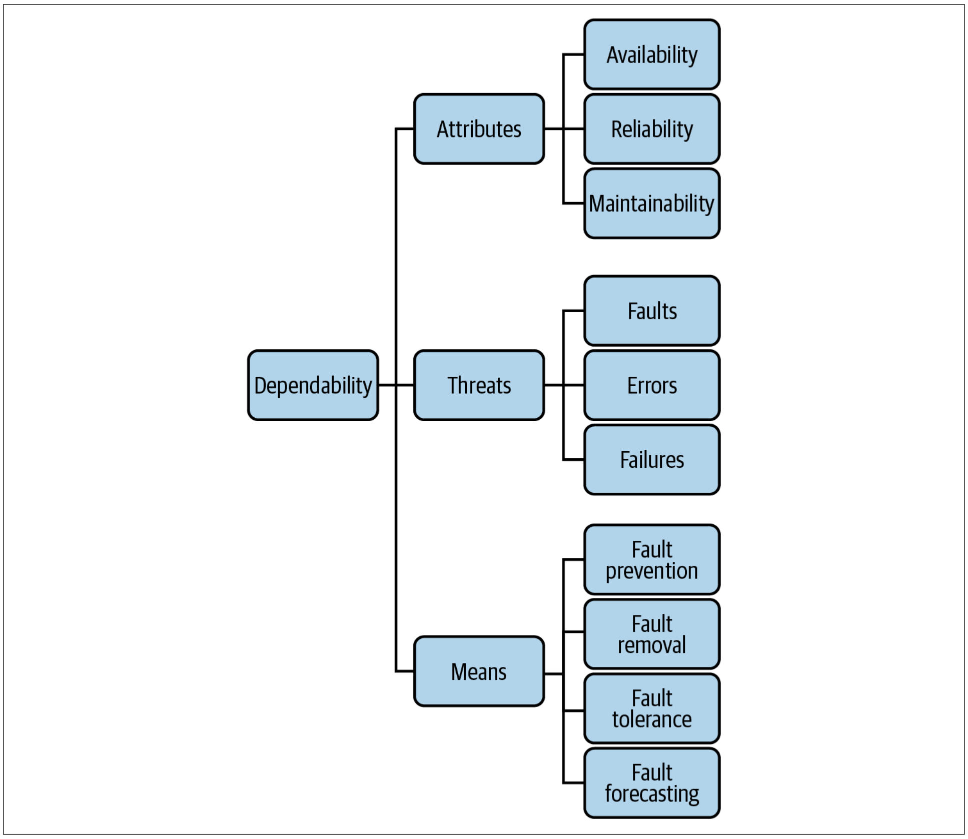


Figure 6-1. The system attributes and means that contribute to dependability

So while the concept of "dependability" alone might be a little squishy and subjective, the attributes that contribute to it are quantitative and measurable enough to be useful:

Availability
: The ability of a system to perform its intended function at a random moment in time. This is usually expressed as the probability that a request made of the system will be successful, defined as uptime divided by total time.

Reliability
: The ability of a system to perform its intended function for a given time interval. This is often expressed as either the mean time between failures (MTBF: total time divided by the number of failures) or failure rate (number of failures divided by total time).

Maintainability

The ability of a system to undergo modifications and repairs. There are a variety of indirect measures for maintainability, ranging from calculations of cyclomatic complexity to tracking the amount of time required to change a system's behavior to meet new requirements or to restore it to a functional state.

Later authors extended Laprie's definition of dependability to include several security-related properties, including safety, confidentiality, and integrity. I've reluctantly omitted these, not because security isn't important (it's *SO* important!), but for brevity. A worthy discussion of security would require an entire book of its own.

Dependability Is Not Reliability

If you've read any of O'Reilly's Site Reliability Engineering (SRE) books[9] you've already heard quite a lot about reliability. However, as illustrated in Figure 6-1, reliability is just one property that contributes to overall dependability.

If that's true, though, then why has reliability become the standard metric for service functionality? Why are there "site reliability engineers" but no "site dependability engineers"?

There are probably several answers to these questions, but perhaps the most definitive is that the definition of "dependability" is purely qualitative. There's no measure for it, and when you can't measure something it's very hard to construct a set of rules around it.

Reliability, on the other hand, is quantitative. Given a robust definition[10] for what it means for a system to provide "correct" service, it becomes relatively straightforward to calculate that system's "reliability," making it a powerful (if indirect) measure of user experience.

Dependability: It's Not Just for Ops Anymore

Since the introduction of networked services, it's been the job of developers to build services, and of systems administrators ("operations") to deploy those services onto servers and keep them running. This worked well enough for a time, but it had the unfortunate side-effect of incentivizing developers to prioritize feature development at the expense of stability and operations.

9 If you haven't, start with *Site Reliability Engineering: How Google Runs Production Systems*. It really is very good.

10 Many organizations use service-level objectives (SLOs) for precisely this purpose.

Fortunately, over the past decade or so—coinciding with the DevOps movement—a new wave of technologies has become available with the potential to completely change the way technologists of all kinds do their jobs.

On the operations side, with the availability of infrastructure and platforms as a service (IaaS/PaaS) and tools like Terraform and Ansible, working with infrastructure has never been more like writing software.

On the development side, the popularization of technologies like containers and serverless functions has given developers an entire new set of "operations-like" capabilities, particularly around virtualization and deployment.

As a result, the once-stark line between software and infrastructure is getting increasingly blurry. One could even argue that with the growing advancement and adoption of infrastructure abstractions like virtualization, container orchestration frameworks like Kubernetes, and software-defined behavior like service meshes, we may even be at the point where they could be said to have merged. Everything is software now.

The ever-increasing demand for service dependability has driven the creation of a whole new generation of cloud native technologies. The effects of these new technologies and the capabilities they provide has been considerable, and the traditional developer and operations roles are changing to suit them. At long last, the silos are crumbling, and, increasingly, the rapid production of dependable, high-quality services is a fully collaborative effort of all of its designers, implementors, and maintainers.

Achieving Dependability

This is where the rubber meets the road. If you've made it this far, congratulations.

So far we've discussed Laprie's definition of "dependability," which can be (very) loosely paraphrased as "happy users," and we've discussed the attributes—availability, reliability, and maintainability—that contribute to it. This is all well and good, but without actionable advice for how to achieve dependability the entire discussion is purely academic.

Laprie thought so too, and defined four broad categories of techniques that can be used together to improve a system's dependability (or which, by their absence, can reduce it):

Fault prevention
: Fault prevention techniques are used during system construction to prevent the occurrence or introduction of faults.

Fault tolerance

Fault tolerance techniques are used during system design and implementation to prevent service failures in the presence of faults.

Fault removal

Fault removal techniques are used to reduce the number and severity of faults.

Fault forecasting

Fault forecasting techniques are used to identify the presence, the creation, and the consequences of faults.

Interestingly, as illustrated in Figure 6-2, these four categories correspond surprisingly well to the five cloud native attributes that we introduced all the way back in Chapter 1.

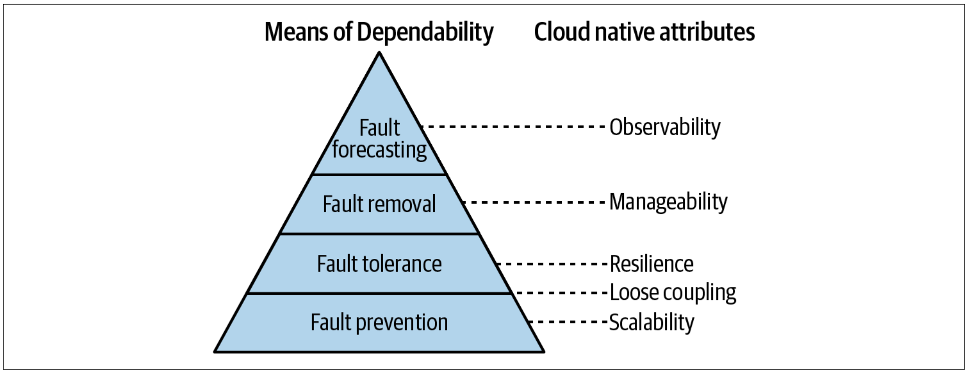


Figure 6-2. The four means of achieving dependability, and their corresponding cloud native attributes

Fault prevention and fault tolerance make up the bottom two layers of the pyramid, corresponding with scalability, loose coupling, and resilience. Designing a system for scalability prevents a variety of faults common among cloud native applications, and resiliency techniques allow a system to tolerate faults when they do inevitably arise. Techniques for loose coupling can be said to fall into both categories, preventing and enhancing a service's fault tolerance. Together these can be said to contribute to what Laprie terms *dependability procurement*: the means by which a system is provided with the ability to perform its designated function.

Techniques and designs that contribute to manageability are intended to produce a system that can be easily modified, simplifying the process of removing faults when they're identified. Similarly, observability naturally contributes to the ability to forecast faults in a system. Together fault removal and forecasting techniques contribute to what Laprie termed *dependability validation*: the means by which confidence is gained in a system's ability to perform its designated function.

Consider the implications of this relationship: what was a purely academic exercise 35 years ago has essentially been rediscovered—apparently independently—as a natural consequence of years of accumulated experience building reliable production systems. Dependability has come full-circle.

In the subsequent sections we'll explore these relationships more fully and preview later chapters, in which we discuss exactly how these two apparently disparate systems actually correspond quite closely.

Fault Prevention

At the base of our "Means of Dependability" pyramid are techniques that focus on preventing the occurrence or introduction of faults. As veteran programmers can attest, many—if not most—classes of errors and faults can be predicted and prevented during the earliest phases of development. As such, many fault prevention techniques come into play during the design and implementation of a service.

Good programming practices

Fault prevention is one of the primary goals of software engineering in general, and is the explicit goal of any development methodology, from pair programming to test-driven development and code review practices. Many such techniques can really be grouped into what might be considered to be "good programming practice," about which innumerable excellent books and articles have already been written, so we won't explicitly cover it here.

Language features

Your choice of language can also greatly affect your ability to prevent or fix faults. Many language features that some programmers have sometimes come to expect, such as dynamic typing, pointer arithmetic, manual memory management, and thrown exceptions (to name a few) can easily introduce unintended behaviors that are difficult to find and fix, and may even be maliciously exploitable.

These kinds of features strongly motivated many of the design decisions for Go, resulting in the strongly typed garbage-collected language we have today. For a refresher for why Go is particularly well suited for the development of cloud native services, take a look back at Chapter 2.

Scalability

We briefly introduced the concept of scalability way back in Chapter 1, where it was defined as the ability of a system to continue to provide correct service in the face of significant changes in demand.

In that section we introduced two different approaches to scaling—vertical scaling (scaling up) by resizing existing resources, and horizontal scaling (scaling out) by adding (or removing) service instances—and some of the pros and cons of each.

We'll go quite a bit deeper into each of these in Chapter 7, especially into the gotchas and downsides. We'll also talk a lot about the problems posed by state.[11] For now, though, it'll suffice to say that having to scale your service adds quite a bit of overhead, including but not limited to cost, complexity, and debugging.

While scaling resources is eventually often inevitable, it's often better (and cheaper!) to resist the temptation to throw hardware at the problem and postpone scaling events as long as possible by considering runtime efficiency and algorithmic scaling. As such, we'll cover a number of Go features and tooling that allow us to identify and fix common problems like memory leaks and lock contention that tend to plague systems at scale.

Loose coupling

Loose coupling, which we first defined in "Loose Coupling" on page 7, is the system property and design strategy of ensuring that a system's components have as little knowledge of other components as possible. The degree of coupling between services can have an enormous—and too often under-appreciated—impact on a system's ability to scale and to isolate and tolerate failures.

Since the beginning of microservices there have been dissenters who point to the difficulty of deploying and maintaining microservice-based systems as evidence that such architectures are just too complex to be viable. I don't agree, but I can see where they're coming from, given how incredibly easy it is to build a *distributed monolith*. The hallmark of a distributed monolith is the tight coupling between its components, which results in an application saddled with all of the complexity of microservices plus the all of the tangled dependencies of the typical monolith. If you have to deploy most of your services together, or if a failed health check sends cascading failures through your entire system, you probably have a distributed monolith.

Building a loosely coupled system is easier said than done, but is possible with a little discipline and reasonable boundaries. In Chapter 8 we'll cover how to use data exchange contracts to establish those boundaries, and different synchronous and asynchronous communication models and architectural patterns and packages used to implement them and avoid the dreaded distributed monolith.

11 Application state is hard, and when done wrong it's poison to scalability.

Fault Tolerance

Fault tolerance has a number of synonyms—self-repair, self-healing, resilience—that all describe a system's ability to detect errors and prevent them from cascading into a full-blown failure. Typically, this consists of two parts: *error detection*, in which an error is discovered during normal service, and *recovery*, in which the system is returned to a state where it can be activated again.

Perhaps the most common strategy for providing resilience is redundancy: the duplication of critical components (having multiple service replicas) or functions (retrying service requests). This is a broad and very interesting field with a number of subtle gotchas that we'll dig into in Chapter 9.

Fault Removal

Fault removal, the third of the four dependability means, is the process of reducing the number and severity of faults—latent software flaws that can cause errors—before they manifest as errors.

Even under ideal conditions, there are plenty of ways that a system can error or otherwise misbehave. It might fail to perform an expected action, or perform the wrong action entirely, perhaps maliciously. Just to make things even more complicated, conditions aren't always—or often—ideal.

Many faults can be identified by testing, which allows you to verify that the system (or at least its components) behaves as expected under known test conditions.

But what about unknown conditions? Requirements change, and the real world doesn't care about your test conditions. Fortunately, with effort, a system can be designed to be manageable enough that its behavior can often be adjusted to keep it secure, running smoothly, and compliant with changing requirements.

We'll briefly discuss these next.

Verification and testing

There are exactly four ways of finding latent software faults in your code: testing, testing, testing, and bad luck.

Yes, I joke, but that's not so far from the truth: if you don't find your software faults, your users will. If you're lucky. If you're not, then they'll be found by bad actors seeking to take advantage of them.

Bad jokes aside, there are two common approaches to finding software faults in development:

Static analysis
: Automated, rule-based code analysis performed without actually executing programs. Static analysis is useful for providing early feedback, enforcing consistent practices, and finding common errors and security holes without depending on human knowledge or effort.

Dynamic analysis
: Verifying the correctness of a system or subsystem by executing it under controlled conditions and evaluating its behavior. More commonly referred to simply as "testing."

Key to software testing is having software that's *designed for testability* by minimizing the *degrees of freedom*—the range of possible states—of its components. Highly testable functions have a single purpose, with well-defined inputs and outputs and few or no *side effects*; that is, they don't modify variables outside of their scope. If you'll forgive the nerdiness, this approach minimizes the *search space*—the set of all possible solutions—of each function.

Testing is a critical step in software development that's all too often neglected. The Go creators understood this and baked unit testing and benchmarking into the language itself in the form of the `go test` command and the testing package. Unfortunately, a deep dive into testing theory is well beyond the scope of this book, but we'll do our best to scratch the surface in Chapter 9.

Manageability

Faults exist when your system doesn't behave according to requirements. But what happens when those requirements change?

Designing for *manageability*, first introduced back in "Manageability" on page 10, allows a system's behavior to be adjusted without code changes. A manageable system essentially has "knobs" that allow real-time control to keep your system secure, running smoothly, and compliant with changing requirements.

Manageability can take a variety of forms, including (but not limited to!) adjusting and configuring resource consumption, applying on-the-fly security remediations, *feature flags* that can turn features on or off, or even loading plug-in-defined behaviors.

Clearly, manageability is a broad topic. We'll review a few of the mechanisms Go provides for it in Chapter 10.

Fault Forecasting

At the peak of our "Means of Dependability" pyramid (Figure 6-2) is *fault forecasting*, which builds on the knowledge gained and solutions implemented in the levels below it to attempt to estimate the present number, the future incidence, and the likely consequence of faults.

Too often this consists of guesswork and gut feelings instead, generally resulting in unexpected failures when a starting assumption stops being true. More systematic approaches include Failure Mode and Effects Analysis and stress testing, which are very useful for understanding a system's possible failure modes.

In a system designed for *observability*, which we'll discuss in depth in Chapter 11, failure mode indicators can be tracked so that they can be forecast and corrected before they manifest as errors. Furthermore, when unexpected failures occur—as they inevitably will—observable systems allow the underlying faults to be quickly identified, isolated, and corrected.

The Continuing Relevance of the Twelve-Factor App

In the early 2010s, developers at Heroku, a platform as a service (PaaS) company and early cloud pioneer, realized that they were seeing web applications being developed again and again with the same fundamental flaws.

Motivated by what they felt were systemic problems in modern application development, they drafted *The Twelve-Factor App*. This was a set of twelve rules and guidelines constituting a development methodology for building web applications, and by extension, cloud native applications (although "cloud native" wasn't a commonly used term at the time). The methodology was for building web applications that: [12]

- Use declarative formats for setup automation, to minimize time and cost for new developers joining the project
- Have a clean contract with the underlying operating system, offering maximum portability between execution environments
- Are suitable for deployment on modern cloud platforms, obviating the need for servers and systems administration

12 Wiggins, Adam. *The Twelve-Factor App*. 2011. *https://12factor.net*.

- Minimize divergence between development and production, enabling continuous deployment for maximum agility
- Can scale up without significant changes to tooling, architecture, or development practices

While not fully appreciated when it was first published in 2011, as the complexities of cloud native development have become more widely understood (and felt), *The Twelve Factor App* and the properties it advocates have started to be cited as the bare minimum for any service to be cloud native.

I. Codebase

> One codebase tracked in revision control, many deploys.
>
> —The Twelve-Factor App

For any given service, there should be exactly one codebase that's used to produce any number of immutable releases for multiple deployments to multiple environments. These environments typically include a production site, and one or more staging and development sites.

Having multiple services sharing the same code tends to lead to a blurring of the lines between modules, trending in time to something like a monolith, making it harder to make changes in one part of the service without affecting another part (or another service!) in unexpected ways. Instead, shared code should be refactored into libraries that can be individually versioned and included through a dependency manager.

Having a single service spread across multiple repositories, however, makes it nearly impossible to automatically apply the build and deploy phases of your service's life cycle.

II. Dependencies

> Explicitly declare and isolate (code) dependencies.
>
> —The Twelve-Factor App

For any given version of the codebase, `go build`, `go test`, and `go run` should be deterministic: they should have the same result, however they're run, and the product should always respond the same way to the same inputs.

But what if a dependency—an imported code package or installed system tool beyond the programmer's control—changes in such a way that it breaks the build, introduces a bug, or becomes incompatible with the service?

Most programming languages offer a packaging system for distributing support libraries, and Go is no different.[13] By using Go modules to declare all dependencies, completely and exactly, you can ensure that imported packages won't change out from under you and break your build in unexpected ways.

To extend this somewhat, services should generally try to avoid using the `os/exec` package's `Command` function to shell out to external tools like ImageMagick or `curl`.

Yes, your target tool might be available on all (or most) systems, but there's no way to *guarantee* that they both exist and are fully compatible with the service everywhere that it might run in the present or future. Ideally, if your service requires an external tool, that tool should be *vendored* into the service by including it in the service's repository.

III. Configuration

> Store configuration in the environment.
>
> —The Twelve-Factor App

Configuration—anything that's likely to vary between environments (staging, production, developer environments, etc)—should always be cleanly separated from the code. Under no circumstances should an application's configuration be baked into the code.

Configuration items may include, but certainly aren't limited to:

- URLs or other resource handles to a database or other upstream service dependencies—even if it's not likely to change any time soon.
- Secrets of *any* kind, such as passwords or credentials for external services.
- Per-environment values, such as the canonical hostname for the deploy.

A common means of extracting configuration from code is by *externalizing* them into some configuration file—often YAML[14]—which may or may not be checked into the repository alongside the code. This is certainly an improvement over configuration-in-code, but it's also less than ideal.

First, if your configuration file lives outside of the repository, it's all too easy to accidentally check it in. What's more, such files tend to proliferate, with different versions for different environments living in different places, making it hard to see and manage configurations with any consistency.

13 Although it was for too long!

14 The world's worst configuration language (except for all the other ones).

Alternatively, you *could* have different versions of your configurations for each environment in the repository, but this can be unwieldy and tends to lead to some awkward repository acrobatics.

There Are No Partially Compromised Secrets

It's worth emphasizing that while configuration values should never be in code, passwords or other sensitive secrets should *absolutely, never ever* be in code. It's all too easy for those secrets, in a moment of forgetfulness, to get shared with the whole world.

Once a secret is out, it's out. There are no partially compromised secrets.

Always treat your repository—and the code it contains—as if it can be made public at any time. Which, of course, it can.

Instead of configurations as code or even as external configurations, *The Twelve Factor App* recommends that configurations be stored as *environment variables*. Using environment variables in this way actually has a lot of advantages:

- They are standard and largely OS and language agnostic.
- That are easy to change between deploys without changing any code.
- They're very easy to inject into containers.

Go has several tools for doing this.

The first—and most basic—is the `os` package, which provides the `os.Getenv` function for this purpose:

```
name := os.Getenv("NAME")
place := os.Getenv("CITY")

fmt.Printf("%s lives in %s.\n", name, place)
```

For more sophisticated configuration options, there are several excellent packages available. Of these, `spf13/viper` seems to be particularly popular. A snippet of Viper in action might look like the following:

```
viper.BindEnv("id")              // Will be uppercased automatically
viper.SetDefault("id", "13")     // Default value is "13"

id1 := viper.GetInt("id")
fmt.Println(id1)                 // 13

os.Setenv("ID", "50")            // Typically done outside of the app!
```

```
    id2 := viper.GetInt("id")
    fmt.Println(id2)                // 50
```

Additionally, Viper provides a number of features that the standard packages do not, such as default values, typed variables, and reading from command-line flags, variously formatted configuration files, and even remote configuration systems like etcd and Consul.

We'll dive more deeply into Viper and other configuration topics in Chapter 10.

IV. Backing Services

> Treat backing services as attached resources.
>
> —The Twelve-Factor App

A backing service is any downstream dependency that a service consumes across the network as part of its normal operation (see "Upstream and Downstream Dependencies" on page 5). A service should make no distinction between backing services of the same type. Whether it's an internal service that's managed within the same organization or a remote service managed by a third party should make no difference.

To the service, each distinct upstream service should be treated as just another resource, each addressable by a configurable URL or some other resource handle, as illustrated oi Figure 6-3. All resources should be treated as equally subject to the *Fallacies of Distributed Computing* (see Chapter 4 for a refresher, if necessary).

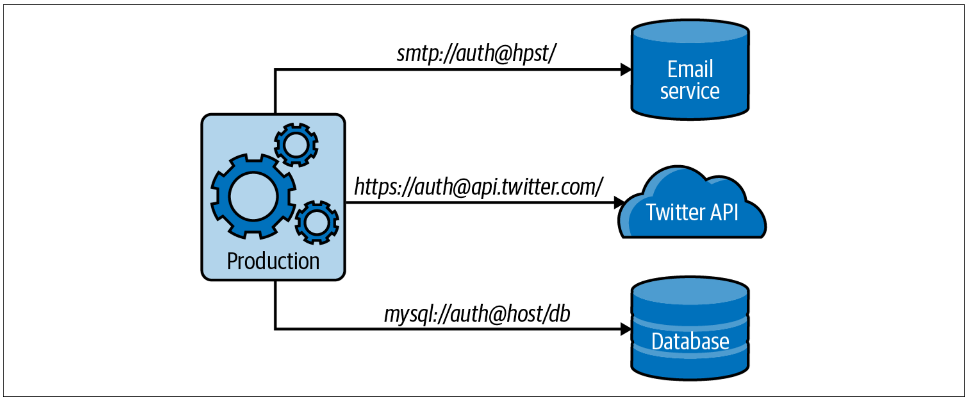


Figure 6-3. Each upstream service should be treated as just another resource, each addressable by a configurable URL or some other resource handle, each equally subject to the Fallacies of Distributed Computing

In other words, a MySQL database run by your own team's sysadmins should be treated no differently than an AWS-managed RDS instance. The same goes for *any* upstream service, whether it's running in a data center in another hemisphere or in a Docker container on the same server.

A service that's able to swap out any resource at will with another one of the same kind—internally managed or otherwise—just by changing a configuration value can be more easily deployed to different environments, can be more easily tested, and more easily maintained.

V. Build, Release, Run

> Strictly separate build and run stages.
>
> —The Twelve-Factor App

Each (nondevelopment) deployment—the union of a specific version of the built code and a configuration—should be immutable and uniquely labeled. It should be possible, if necessary, to precisely recreate a deployment if (heaven forbid) it is necessary to roll a deployment back to an earlier version.

Typically, this is accomplished in three distinct stages, illustrated in Figure 6-4 and described in the following:

Build
: In the build stage, an automated process retrieves a specific version of the code, fetches dependencies, and compiles an executable artifact we call a *build*. Every build should always have a unique identifier, typically a timestamp or an incrementing build number.

Release
: In the release stage, a specific build is combined with a configuration specific to the target deployment. The resulting *release* is ready for immediate execution in the execution environment. Like builds, releases should also have a unique identifier. Importantly, producing releases with same version of a build shouldn't involve a rebuild of the code: to ensure environment parity, each environment-specific configuration should use the same build artifact.

Run
: In the run stage, the release is delivered to the deployment environment and executed by launching the service's processes.

Ideally, a new versioned build will be automatically produced whenever new code is deployed.

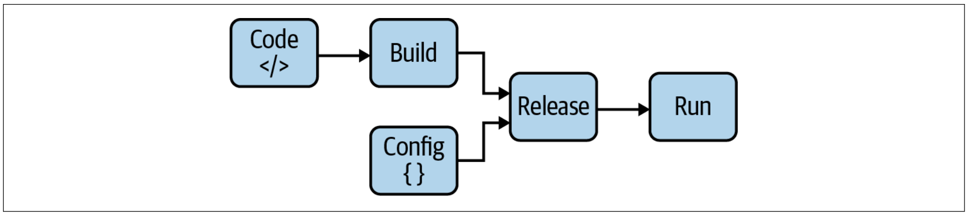


Figure 6-4. The process of deploying a codebase to a (nondevelopment) environment should be performed in distinct build, release, and run stages

VI. Processes

> Execute the app as one or more stateless processes.
>
> —The Twelve-Factor App

Service processes should be stateless and share nothing. Any data that has to be persisted should be stored in a stateful backing service, typically a database or external cache.

We've already spent some time talking about statelessness—and we'll spend more in the next chapter—so we won't dive into this point any further.

However, if you're interested in reading ahead, feel free to take a look at "State and Statelessness" on page 195.

VII. Data Isolation

> Each service manages its own data.
>
> —Cloud Native, *Data Isolation*

Each service should be entirely *self-contained*. That is, it should manage its own data, and make its data accessible only via an API designed for that purpose. If this sounds familiar to you, good! This is actually one of the core principles of microservices, which we'll discuss more in "The Microservices System Architecture" on page 211.

Very often this will be implemented as a request-response service like a RESTful API or RPC protocol that's exported by listening to requests coming in on a port, but this can also take the form of an asynchronous, event-based service using a publish-subscribe messaging pattern. Both of these patterns will be described in more detail in Chapter 8.

Historical Note

The actual title of the seventh section of *The Twelve Factor App* is “Port Binding,” and is summarized as “export services via port binding.”[15]

At the time, this advice certainly made sense, but this title obscures its main point: that a service should encapsulate and manage its own data, and only share that data via an API.

While many (or even most) web applications do, in fact, expose their APIs via ports, the increasing popularity of functions as a service (FaaS) and event-driven architectures means this is no longer necessarily always the case.

So, instead of the original text, I’ve decided to use the more up-to-date (and true-to-intent) summary provided by Boris Scholl, Trent Swanson, and Peter Jausovec in *Cloud Native: Using Containers, Functions, and Data to Build Next-Generation Applications* (O’Reilly).

And finally, although this is something you don’t see in the Go world, some languages and frameworks allow the runtime injection of an application server into the execution environment to create a web-facing service. This practice limits testability and portability by breaking data isolation and environment agnosticism, and is *very strongly* discouraged.

VIII. Scalability

> Scale out via the process model.
>
> —The Twelve-Factor App

Services should be able to scale horizontally by adding more instances.

We talk about scalability quite a bit in this book. We even dedicated all of Chapter 7 to it. With good reason: the importance of scalability can’t be understated.

Sure, it’s certainly convenient to just beef up the one server your service is running on —and that’s fine in the (very) short term—but vertical scaling is a losing strategy in the long run. If you’re lucky, you’ll eventually hit a point where you simply can’t scale up any more. It’s more likely that your single server will either suffer load spikes faster than you can scale up, or just die without warning and without a redundant failover.[16] Both scenarios end with a lot of unhappy users. We’ll discuss scalability quite a bit more in Chapter 7.

15 Wiggins, Adam. “Port Binding.” *The Twelve-Factor App*. 2011. *https://oreil.ly/bp8lC*.

16 Probably at three in the morning.

IX. Disposability

Maximize robustness with fast startup and graceful shutdown.

—The Twelve-Factor App

Cloud environments are fickle: provisioned servers have a funny way of disappearing at odd times. Services should account for this by being *disposable*: service instances should be able to be started or stopped—intentionally or not—at any time.

Services should strive to minimize the time it takes to start up to reduce the time it takes for the service to be deployed (or redeployed) to elastically scale. Go, having no virtual machine or other significant overhead, is especially good at this.

Containers provide fast startup time and are also very useful for this, but care must be taken to keep image sizes small to minimize the data transfer overhead incurred with each initial deploy of a new image. This is another area in which Go excels: its self-sufficient binaries can generally be installed into `SCRATCH` images, without requiring an external language runtime or other external dependencies. We demonstrated this in the previous chapter, in "Containerizing Your Key-Value Store" on page 150.

Services should also be capable of shutting down when they receive a `SIGTERM` signal by saving all data that needs to be saved, closing open network connections, or finishing any in-progress work that's left or by returning the current job to the work queue.

X. Development/Production Parity

Keep development, staging, and production as similar as possible.

—The Twelve-Factor App

Any possible differences between development and production should be kept as small as possible. This includes code differences, of course, but it extends well beyond that:

Code divergence
: Development branches should be small and short-lived, and should be tested and deployed into production as quickly as possible. This minimizes functional differences between environments and reduces the risk of both deploys and rollbacks.

Stack divergence
: Rather than having different components for development and production (say, SQLite on OS X versus MySQL on Linux), environments should remain as similar as possible. Lightweight containers are an excellent tool for this. This minimizes the possibility that inconvenient differences between almost-but-not-quite-the-same implementations will emerge to ruin your day.

Personnel divergence

Once it was common to have programmers who wrote code and operators who deployed code, but that arrangement created conflicting incentives and counter-productive adversarial relationships. Keeping code authors involved in deploying their work and responsible for its behavior in production helps break down development/operations silos and aligns incentives around stability and velocity.

Taken together, these approaches help to keep the gap between development and production small, which in turn encourages rapid, automated, continuous deployment.

XI. Logs

Treat logs as event streams.

—The Twelve-Factor App

Logs—a service's never-ending stream of consciousness—are incredibly useful things, particularly in a distributed environment. By providing visibility into the behavior of a running application, good logging can greatly simplify the task of locating and diagnosing misbehavior.

Traditionally, services wrote log events to a file on the local disk. At cloud scale, however, this just makes valuable information awkward to find, inconvenient to access, and impossible to aggregate. In dynamic, ephemeral environments like Kubernetes your service instances (and their log files) may not even exist by the time you get around to viewing them.

Instead, a cloud native service should treat log information as nothing more than a stream of events, writing each event, unbuffered, directly to `stdout`. It shouldn't concern itself with implementation trivialities like routing or storage of its log events, and allow the executor to decide what happens to them.

Though seemingly simple (and perhaps somewhat counterintuitive), this small change provides a great deal of freedom.

During local development, a programmer can watch the event stream in a terminal to observe the service's behavior. In deployment, the output stream can be captured by the execution environment and forwarded to one or more destinations, such as a log indexing system like Elasticsearch, Logstash, and Kibana (ELK) or Splunk for review and analysis, or a data warehouse for long-term storage.

We'll discuss logs and logging, in the context of observability, in more detail in Chapter 11.

XII. Administrative Processes

> Run administrative/management tasks as one-off processes.
>
> —The Twelve-Factor App

Of all of the original Twelve Factors, this is the one that most shows its age. For one thing, it explicitly advocates shelling into an environment to manually execute tasks.

To be clear: *making manual changes to a server instance creates snowflakes. This is a bad thing.* See "Special Snowflakes" on page 188.

Assuming you even have an environment that you can shell into, you should assume that it can (and eventually will) be destroyed and re-created any moment.

Ignoring all of that for a moment, let's distill the point to its original intent: administrative and management tasks should be run as one-off processes. This could be interpreted in two ways, each requiring its own approach:

- If your task is an administrative process, like a data repair job or database migration, it should be run as a short-lived process. Containers and functions are excellent vehicles for such purposes.
- If your change is an update to your service or execution environment, you should instead modify your service or environment construction/configuration scripts, respectively.

Special Snowflakes

Keeping servers healthy can be a challenge. At 3 a.m., when things aren't working quite right, it's really tempting to make a quick change and go back to bed. Congratulations, you've just created a *snowflake*: a special server instance with manual changes that give it unique, usually undocumented, behaviors.

Even minor, seemingly harmless changes can lead to significant problems. Even if the changes are documented—which is rarely the case—snowflake servers are hard to reproduce exactly, particularly if you need to keep an entire cluster in sync. This can lead to a bad time when you have to redeploy your service onto new hardware and can't figure out why it's not working.

Furthermore, because your testing environment no longer matches production, you can no longer trust your development environments to reliably reproduce your production deployment.

Instead, servers and containers should be treated as *immutable*. If something needs to be updated, fixed, or modified in any way, changes should be made by updating the appropriate build scripts, baking[17] a new common image, and provisioning new server or container instances to replace the old ones.

As the expression goes, instances should be treated as "cattle, not pets."

Summary

In this chapter we considered the question "what's the point of cloud native?" The common answer is "a computer system that works in the cloud." But "work" can mean anything. Surely we can do better.

So we went back to thinkers like Tony Hoare and J-C Laprie, who provided the first part of the answer: *dependability*. That is, to paraphrase, computer systems that behave in ways that users find acceptable, despite living in a fundamentally unreliable environment.

Obviously, that's more easily said than done, so we reviewed three schools of thought regarding how to achieve it:

- Laprie's academic "means of dependability," which include preventing, tolerating, removing, and forecasting faults
- Adam Wiggins' *Twelve Factor App*, which took a more prescriptive (and slightly dated, in spots) approach
- Our own "cloud native attributes," based on the Cloud Native Computing Foundation's definition of "cloud native," that we introduced in Chapter 1 and organized this entire book around

Although this chapter was essentially a short survey of theory, there's a lot of important, foundational information here that describes the motivations and means used to achieve what we call "cloud native."

17 "Baking" is a term sometimes used to refer to the process of creating a new container or server image.

CHAPTER 7

Scalability

Some of the best programming is done on paper, really. Putting it into the computer is just a minor detail.[1]

—Max Kanat-Alexander, *Code Simplicity: The Fundamentals of Software*

In the summer of 2016, I joined a small company that digitized the kind of forms and miscellaneous paperwork that state and local governments are known and loved for. The state of their core application was pretty typical of early-stage startups, so we got to work and, by that fall, had managed to containerize it, describe its infrastructure in code, and fully automate its deployment.

One of our clients was a small coastal city in southeastern Virginia, so when Hurricane Matthew—the first Category 5 Atlantic hurricane in nearly a decade—was forecast to make landfall not far from there, the local officials dutifully declared a state of emergency and used our system to create the necessary paperwork for citizens to fill out. Then they posted it to social media, and half a million people all logged in at the same time.

When the pager went off, the on-call checked the metrics and found that aggregated CPU for the servers was pegged at 100%, and that hundreds of thousands of requests were timing out.

So, we added a zero to the desired server count, created a "to-do" task to implement autoscaling, and went back to our day. Within 24 hours, the rush had passed, so we scaled the servers in.

What did we learn from this, other than the benefits of autoscaling?[2]

1 Kanat-Alexander, Max. *Code Simplicity: The Science of Software Design*. O'Reilly Media, 23 March 2012.

2 Honestly, if we had autoscaling in place I probably wouldn't even remember that this happened.

First of all, it underscored the fact that without the ability to scale, our system would have certainly suffered extended downtime. But being able to add resources on demand meant that we could serve our users even under load far beyond what we had ever anticipated. As an added benefit, if any one server failed, its work could have been divided among the survivors.

Second, having far more resources than necessary isn't just wasteful, it's expensive. The ability to scale our instances back in when demand ebbed meant that we were only paying for the resources that we needed. A major plus for a startup on a budget.

Unfortunately, because unscalable services can seem to function perfectly well under initial conditions, scalability isn't always a consideration during service design. While this might be perfectly adequate in the short term, services that aren't capable of growing much beyond their original expectations also have a limited lifetime value. What's more, it's often fiendishly difficult to refactor a service for scalability, so building with it in mind can save both time and money in the long run.

First and foremost, this is meant to be a Go book, or at least more of a Go book than an infrastructure or architecture book. While we will discuss things like scalable architecture and messaging patterns, much of this chapter will focus on demonstrating how Go can be used to produce services that lean on the other (non-infrastructure) part of the scalability equation: efficiency.[3]

What Is Scalability?

You may recall that concept of scalability was first introduced way back in Chapter 1, where it was defined as the ability of a system to continue to provide correct service in the face of significant changes in demand. By this definition, a system can be considered to be scalable if it doesn't need to be redesigned to perform its intended function during steep increases in load.

Note that this definition[4] doesn't actually say anything at all about adding physical resources. Rather, it calls out a system's ability to handle large swings in demand. The thing being "scaled" here is the magnitude of the demand. While adding resources is one perfectly acceptable means of achieving scalability, it isn't exactly the same as being scalable. To make things just a little more confusing, the word "scaling" can also be applied to a system, in which case it *does* mean a change in the amount of dedicated resources.

3 If you want to know more about cloud native infrastructure and architecture, a bunch of excellent books on the subject have already been written. I particularly recommend *Cloud Native Infrastructure* by Justin Garrison and Kris Nova, and *Cloud Native Transformation* by Pini Reznik, Jamie Dobson, and Michelle Gienow (both O'Reilly Media).

4 This is my definition. I acknowledge that it diverges from other common definitions.

So how do we handle high demand without adding resources? As we'll discuss in "Scaling Postponed: Efficiency" on page 197, systems built with *efficiency* in mind are inherently more scalable by virtue of their ability to gracefully absorb high levels of demand, without immediately having to resort to adding hardware in response to every dramatic swing in demand, and without having to massively over-provision "just in case."

Different Forms of Scaling

Unfortunately, even the most efficient of efficiency strategies has its limit, and eventually you'll find yourself needing to scale your service to provide additional resources. There are two different ways that this can be done (see Figure 7-1), each with its own associated pros and cons:

Vertical scaling
: A system can be *vertically scaled* (or *scaled up*) by increasing its resource allocations. In a public cloud, an existing server can be vertically scaled fairly easily just by changing its instance size, but only until you run out of larger instance types (or money).

Horizontal scaling
: A system can be *horizontally scaled* (or *scaled out*) by duplicating the system or service to limit the burden on any individual server. Systems using this strategy can typically scale to handle greater amounts of load, but as you'll see in "State and Statelessness" on page 195, the presence of state can make this strategy difficult or impossible for some systems.

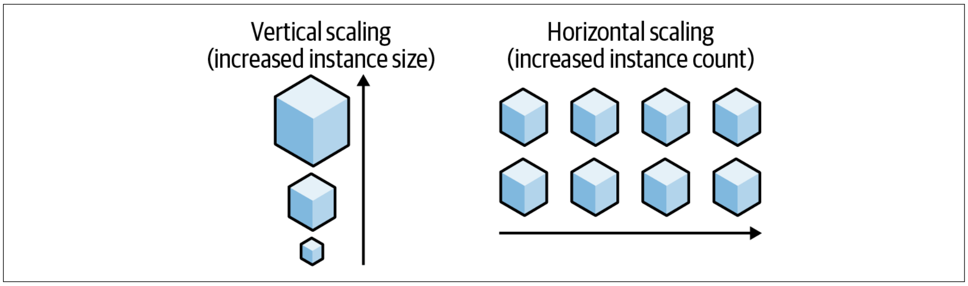


Figure 7-1. Vertical scaling can be an effective short-term solution; horizontal scaling is more technically challenging but may be a better long-term strategy

These two terms are used to describe the most common way of thinking about scaling: taking an entire system, and just making *more* of it. There are a variety of other scaling strategies in use, however.

Perhaps the most common of these is *functional partitioning*, which you're no doubt already familiar with, if not by name. Functional partitioning involves decomposing complex systems into smaller functional units that can be independently optimized, managed, and scaled. You might recognize this as a generalization of a number of best practices ranging from basic program design to advanced distributed systems design.

Another approach common in systems with large amounts of data—particularly databases—is *sharding*. Systems that use this strategy distribute load by dividing their data into partitions called *shards*, each of which holds a specific subset of the larger dataset. A basic example of this is presented in "Minimizing locking with sharding" on page 205.

The Four Common Bottlenecks

As the demands on a system increase, there will inevitably come a point at which one resource just isn't able to keep pace, effectively stalling any further efforts to scale. That resource has become a *bottleneck*.

Returning the system to operable performance levels requires identifying and addressing the bottleneck. This can be done in the short-term by increasing the bottlenecked component—vertically scaling—by adding more memory or up-sizing the CPU, for instance. As you might recall from the discussion in "Different Forms of Scaling" on page 193, this approach isn't always possible (or cost-effective), and it can never be relied upon forever.

However, it's often possible to address a bottleneck by enhancing or reducing the burden on the affected component by utilizing another resource that the system still has in abundance. A database might avoid disk I/O bottlenecking by caching data in RAM; conversely a memory-hungry service could page data to disk. Horizontally scaling doesn't make a system immune: adding more instances can mean more communication overhead, which puts additional strain on the network. Even highly-concurrent systems can become victims of their own inner workings as the demand on them increases, and phenomena like lock contention come into play. Using resources effectively often means making tradeoffs.

Of course, fixing a bottleneck requires that you first identify the constrained component, and while there are many different resources that can emerge as targets for scaling efforts—whether by actually scaling the resource or by using it more efficiently—such efforts tend to focus on just four resources:

CPU
: The number of operations per unit of time that can be performed by a system's central processor and a common bottleneck for many systems. Scaling strategies for CPU include caching the results of expensive deterministic operations (at the expense of memory), or simply increasing the size or number of processors (at the expense of network I/O if scaling out).

Memory

The amount of data that can be stored in main memory. While today's systems can store incredible amounts of data on the order of tens or hundreds of gigabytes, even this can fall short, particularly for data-intensive systems that lean on memory to circumvent disk I/O speed limits. Scaling strategies include offloading data from memory to disk (at the expense of disk I/O) or an external dedicated cache (at the expense of network I/O), or simply increasing the amount of available memory.

Disk I/O

The speed at which data can be read from and written to a hard disk or other persistent storage medium. Disk I/O is a common bottleneck on highly parallel systems that read and write heavily to disk, such as databases. Scaling strategies include caching data in RAM (at the expense of memory) or using an external dedicated cache (at the expense of network I/O).

Network I/O

The speed at which data can be sent across a network, either from a particular point or in aggregate. Network I/O translates directly into *how much* data the network can transmit per unit of time. Scaling strategies for network I/O are often limited,[5] but network I/O is particularly amenable to various optimization strategies that we'll discuss shortly.

As the demand on a system increases, it'll almost certainly find itself bottlenecked by one of these, and while there are efficiency strategies that can be applied, those tend to come at the expense of one or more other resources, so you'll eventually find your system being bottlenecked *again* by another resource.

State and Statelessness

We briefly touched on statelessness in "Application State Versus Resource State" on page 124, where we described application state—server-side data about the application or how it's being used by a client—as something to be avoided if at all possible. But this time, let's spend a little more time discussing what state is, why it can be problematic, and what we can do about it.

It turns out that "state" is strangely difficult to define, so I'll do my best on my own. For the purposes of this book I'll define state as the set of an application's variables which, if changed, affect the behavior of the application.[6]

5 Some cloud providers impose lower network I/O limits on smaller instances. Increasing the size of the instance may increase these limits in some cases.

6 If you have a better definition, let me know. I'm already thinking about the second edition.

Application State Versus Resource State

Most applications have some form of state, but not all state is created equal. It comes in two kinds, one of which is far less desirable than the other.

First, there's *application state*, which exists any time an application needs to remember an event locally. Whenever somebody talks about a *stateful* application, they're usually talking about an application that's designed to use this kind of local state. "Local" is an operative word here.

Second, there's *resource state*, which is the same for every client and which has nothing to do with the actions of clients, like data stored in external data store or managed by configuration management. It's misleading, but saying that an application is *stateless* doesn't mean that it doesn't have any data, just that it's been designed in such a way that it's free of any local persistent data. Its only state is resource state, often because all of its state is stored in some external data store.

To illustrate the difference between the two, imagine an application that tracks client sessions, associating them with some application context. If users' session data was maintained locally by the application, that would be considered "application state." But if the data was stored in an external database, then it could be treated as a remote resource, and it would be "resource state."

Application state is something of the "anti-scalability." Multiple instances of a stateful service will quickly find their individual states diverging due to different inputs being received by each replica. Server affinity provides a workaround to this specific condition by ensuring that each of a client's requests are made to the same server, but this strategy poses a considerable data risk, since the failure of any single server is likely to result in a loss of data.

Advantages of Statelessness

So far, we've discussed the differences between application state and resource state, and we've even suggested—without much evidence (yet)—that application state is bad. However, statelessness provides some very noticeable advantages:

Scalability
: The most visible and most often cited benefit is that stateless applications can handle each request or interaction independent of previous requests. This means that any service replica can handle any request, allowing applications to grow, shrink, or be restarted without losing data required to handle any in-flight sessions or requests. This is especially important when autoscaling your service, because the instances, nodes, or pods hosting the service can (and usually will) be created and destroyed unexpectedly.

Durability
: Data that lives in exactly one place (such as a single service replica) can (and, at some point, *will*) get lost when that replica goes away for any reason. Remember: everything in "the cloud" evaporates eventually.

Simplicity
: Without any application state, stateless services are freed from the need to… well… manage their state.[7] Not being burdened with having to maintain service-side state synchronization, consistency, and recovery logic[8] makes stateless APIs less complex, and therefore easier to design, build, and maintain.

Cacheability
: APIs provided by stateless services are relatively easy to design for cacheability. If a service knows that the result of a particular request will always be the same, regardless of who's making it or when, the result can be safely set aside for easy retrieval later, increasing efficiency and reducing response time.

These might seem like four different things, but there's overlap with respect to what they provide. Specifically, statelessness makes services both simpler and safer to build, deploy, and maintain.

Scaling Postponed: Efficiency

In the context of cloud computing, we usually think of scalability in terms of the ability of a system to add network and computing resources. Often neglected, however, is the role of *efficiency* in scalability. Specifically, the ability for a system to handle changes in demand *without* having to add (or greatly over-provision) dedicated resources.

While it can be argued that most people don't care about program efficiency most of the time, this starts to become less true as demand on a service increases. If a language has a relatively high per-process concurrency overhead—often the case with dynamically typed languages—it will consume all available memory or compute resources much more quickly than a lighter-weight language, and consequently require resources and more scaling events to support the same demand.

This was a major consideration in the design of Go's concurrency model, whose goroutines aren't threads at all but lightweight routines multiplexed onto multiple OS threads. Each costs little more than the allocation of stack space, allowing potentially millions of concurrently executing routines to be created.

7 I know I said the word "state" a bunch of times there. Writing is hard.

8 See also: idempotence.

As such, in this section we'll cover a selection of Go features and tooling that allow us to avoid common scaling problems, such as memory leaks and lock contention, and to identify and fix them when they do arise.

Efficient Caching Using an LRU Cache

Caching to memory is a very flexible efficiency strategy that can be used to relieve pressure on anything from CPU to disk I/O or network I/O, or even just to reduce latency associated with remote or otherwise slow-running operations.

The concept of caching certainly *seems* straightforward. You have something you want to remember the value of—like the result of an expensive (but deterministic) calculation—and you put it in a map for later. Right?

Well, you could do that, but you'll soon start running into problems. What happens as the number of cores and goroutines increases? Since you didn't consider concurrency, you'll soon find your modifications stepping on one another, leading to some unpleasant results. Also, since we forgot to remove anything from our map, it'll continue growing indefinitely until it consumes all of our memory.

What we need is a cache that:

- Supports concurrent read, write, and delete operations
- Scales well as the number of cores and goroutines increase
- Won't grow without limit to consume all available memory

One common solution to this dilemma is an LRU (Least Recently Used) cache: a particularly lovely data structure that tracks how recently each of its keys have been "used" (read or written). When a value is added to the cache such that it exceeds a predefined capacity, the cache is able to "evict" (delete) its least recently used value.

A detailed discussion of how to implement an LRU cache is beyond the scope of this book, but I will say that it's quite clever. As illustrated on Figure 7-2, an LRU cache contains a doubly linked list (which actually contains the values), and a map that associates each key to a node in the linked list. Whenever a key is read or written, the appropriate node is moved to the bottom of the list, such that the least recently used node is always at the top.

There are a couple of Go LRU cache implementations available, though none in the core libraries (yet). Perhaps the most common can be found as part of the golang/groupcache library. However, I prefer HashiCorp's open source extension to `groupcache`, `hashicorp/golang-lru`, which is better documented and includes `sync.RWMutexes` for concurrency safety.

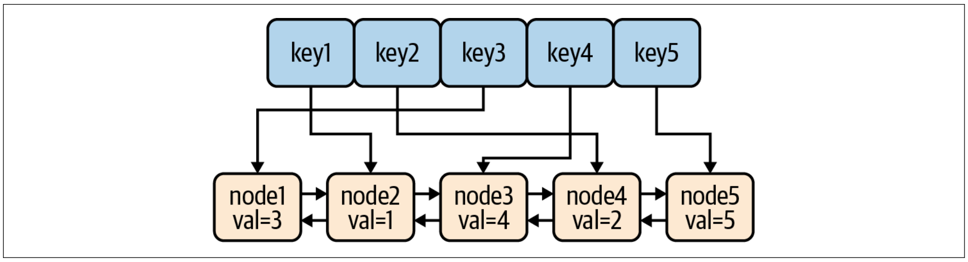


Figure 7-2. An LRU cache contains a map and a doubly linked list, which allows it to discard stale items when it exceeds its capacity

HashiCorp's library contains two construction functions, each of which returns a pointer of type `*Cache` and an `error`:

```
// New creates an LRU cache with the given capacity.
func New(size int) (*Cache, error)

// NewWithEvict creates an LRU cache with the given capacity, and also accepts
// an "eviction callback" function that's called when an eviction occurs.
func NewWithEvict(size int,
    onEvicted func(key interface{}, value interface{})) (*Cache, error)
```

The `*Cache` struct has a number of attached methods, the most useful of which are as follows:

```
// Add adds a value to the cache and returns true if an eviction occurred.
func (c *Cache) Add(key, value interface{}) (evicted bool)

// Check if a key is in the cache (without updating the recent-ness).
func (c *Cache) Contains(key interface{}) bool

// Get looks up a key's value and returns (value, true) if it exists.
// If the value doesn't exist, it returns (nil, false).
func (c *Cache) Get(key interface{}) (value interface{}, ok bool)

// Len returns the number of items in the cache.
func (c *Cache) Len() int

// Remove removes the provided key from the cache.
func (c *Cache) Remove(key interface{}) (present bool)
```

There are several other methods as well. Take a look at the GoDocs for a complete list.

In the following example, we create and use an LRU cache with a capacity of two. To better highlight evictions, we include a callback function that prints some output to `stdout` whenever an eviction occurs. Note that we've decided to initialize the `cache` variable in an `init` function, a special function that's automatically called before the `main` function and after the variable declarations have evaluated their initializers:

```go
package main

import (
    "fmt"
    lru "github.com/hashicorp/golang-lru"
)

var cache *lru.Cache

func init() {
    cache, _ = lru.NewWithEvict(2,
        func(key interface{}, value interface{}) {
            fmt.Printf("Evicted: key=%v value=%v\n", key, value)
        },
    )
}

func main() {
    cache.Add(1, "a")           // adds 1
    cache.Add(2, "b")           // adds 2; cache is now at capacity

    fmt.Println(cache.Get(1))   // "a true"; 1 now most recently used

    cache.Add(3, "c")           // adds 3, evicts key 2

    fmt.Println(cache.Get(2))   // "<nil> false" (not found)
}
```

In the preceding program, we create a `cache` with a capacity of two, which means that the addition of a third value will force the eviction of the least recently used value.

After adding the values `{1:"a"}` and `{2:"b"}` to the cache, we call `cache.Get(1)`, which makes `{1:"a"}` more recently used than `{2:"b"}`. So, when we add `{3:"c"}` in the next step, `{2:"b"}` is evicted, so the next `cache.Get(2)` shouldn't return a value.

If we run this program we'll be able to see this in action. We'll expect the following output:

```
$ go run lru.go
a true
Evicted: key=2 value=b
<nil> false
```

The LRU cache is an excellent data structure to use as a global cache for most use cases, but it does have a limitation: at very high levels of concurrency—on the order of several million operations per second—it will start to experience some contention.

Unfortunately, at the time of this writing, Go still doesn't seem to have a *very* high throughput cache implementation.[9]

Efficient Synchronization

A commonly repeated Go proverb is "don't communicate by sharing memory; share memory by communicating." In other words, channels are generally preferred over shared data structures.

This is a pretty powerful concept. After all, Go's concurrency primitives—goroutines and channels—provide a powerful and expressive synchronization mechanism, such that a set of goroutines using channels to exchange references to data structures can often allow locks to be dispensed with altogether.

(If you're a bit fuzzy on the details of channels and goroutines, don't stress. Take a moment to flip back to "Goroutines" on page 64. It's okay. I'll wait.)

That being said, Go *does* provide more traditional locking mechanisms by way of the `sync` package. But if channels are so great, why would we want to use something like a `sync.Mutex`, and when would we use it?

Well, as it turns out, channels *are* spectacularly useful, but they're not the solution to every problem. Channels shine when you're working with many discrete values, and are the better choice for passing ownership of data, distributing units of work, or communicating asynchronous results. Mutexes, on the other hand, are ideal for synchronizing access to caches or other large stateful structures.

At the end of the day, no tool solves every problem. Ultimately, the best option is to use whichever is most expressive and/or most simple.

Share memory by communicating

Threading is easy; locking is hard.

In this section we're going to use a classic example—originally presented in Andrew Gerrand's classic *Go Blog* article "Share Memory By Communicating"[10]—to demonstrate this truism and show how Go channels can make concurrency safer and easier to reason about.

9 However, if you're interested in learning more about high-performance caching in Go, take a look at Manish Rai Jain's excellent post on the subject, "The State of Caching in Go," on the *Dgraph Blog*.

10 Gerrand, Andrew. "Share Memory By Communicating." *The Go Blog*, 13 July 2010. *https://oreil.ly/GTURp* Portions of this section are modifications based on work created and shared by Google and used according to terms described in the Creative Commons 4.0 Attribution License.

Imagine, if you will, a hypothetical program that polls a list of URLs by sending it a GET request and waiting for the response. The catch is that each request can spend quite a bit of time waiting for the service to respond: anywhere from milliseconds to seconds (or more), depending on the service. Exactly the kind of operation that can benefit from a bit of concurrency, isn't it?

In a traditional threading environment that depends on locking for synchronization you might structure its data something like the following:

```
type Resource struct {
    url        string
    polling    bool
    lastPolled int64
}

type Resources struct {
    data []*Resource
    lock *sync.Mutex
}
```

As you can see, instead of having a slice of URL strings, we have two structs—`Resource` and `Resources`—each of which is already saddled with a number of synchronization structures beyond the URL strings we really care about.

To multithread the polling process in the traditional way, you might have a `Poller` function like the following running in multiple threads:

```
func Poller(res *Resources) {
    for {
        // Get the least recently polled Resource and mark it as being polled
        res.lock.Lock()

        var r *Resource

        for _, v := range res.data {
            if v.polling {
                continue
            }
            if r == nil || v.lastPolled < r.lastPolled {
                r = v
            }
        }

        if r != nil {
            r.polling = true
        }

        res.lock.Unlock()

        if r == nil {
            continue
        }

        // Poll the URL

        // Update the Resource's polling and lastPolled
        res.lock.Lock()
        r.polling = false
        r.lastPolled = time.Nanoseconds()
        res.lock.Unlock()
    }
}
```

This does the job, but it has a lot of room for improvement. It's about a page long, hard to read, hard to reason about, and doesn't even include the URL polling logic or gracefully handle exhaustion of the `Resources` pool.

Now let's take a look at the same functionality implemented using Go channels. In this example, `Resource` has been reduced to its essential component (the URL string), and `Poller` is a function that receives `Resource` values from an input channel, and sends them to an output channel when they're done:

```
type Resource string

func Poller(in, out chan *Resource) {
    for r := range in {
        // Poll the URL

        // Send the processed Resource to out
        out <- r
    }
}
```

It's so…simple. We've completely shed the clockwork locking logic in `Poller`, and our `Resource` data structure no longer contains bookkeeping data. In fact, all that's left are the important parts.

But what if we wanted more than one `Poller` process? Isn't that what we were trying to do in the first place? The answer is, once again, gloriously simple: goroutines. Take a look at the following:

```
for i := 0; i < numPollers; i++ {
    go Poller(in, out)
}
```

By executing `numPollers` goroutines, we're creating `numPollers` concurrent processes, each reading from and writing to the same channels.

A lot has been omitted from the previous examples to highlight the relevant bits. For a walkthrough of a complete, idiomatic Go program that uses these ideas, see the "Share Memory By Communicating" Codewalk.

Reduce blocking with buffered channels

At some point in this chapter you've probably thought to yourself, "sure, channels are great, but writing to channels still blocks." After all, every send operation on a channel blocks until there's a corresponding receive, right? Well, as it turns out, this is only *mostly* true. At least, it's true of default, unbuffered channels.

However, as we first describe in "Channel buffering" on page 65, it's possible to create channels that have an internal message buffer. Send operations on such buffered channels only block when the buffer is full and receives from a channel only block when the buffer is empty.

You may recall that buffered channels can be created by passing an additional capacity parameter to the `make` function to specify the size of the buffer:

```
ch := make(chan type, capacity)
```

Buffered channels are especially useful for handling "bursty" loads. In fact, we already used this strategy in Chapter 5 when we initialized our `FileTransactionLogger`. Distilling some of the logic that's spread through that chapter produces something like the following:

```
type FileTransactionLogger struct {
    events       chan<- Event       // Write-only channel for sending events
    lastSequence uint64             // The last used event sequence number
}

func (l *FileTransactionLogger) WritePut(key, value string) {
    l.events <- Event{EventType: EventPut, Key: key, Value: value}
}

func (l *FileTransactionLogger) Run() {
    l.events = make(chan Event, 16)             // Make an events channel

    go func() {
        for e := range events {                 // Retrieve the next Event
            l.lastSequence++                    // Increment sequence number
        }
    }()
}
```

In this segment, we have a `WritePut` function that can be called to send a message to an `events` channel, which is received in the `for` loop inside the goroutine created in the `Run` function. If `events` was a standard channel, each send would block until the anonymous goroutine completed a receive operation. That might be fine most of the time, but if several writes came in faster than the goroutine could process them, the upstream client would be blocked.

By using a buffered channel we made it possible for this code to handle small bursts of up to 16 closely clustered write requests. Importantly, however, the 17th write *would* block.

It's also important to consider that using buffered channels like this creates a risk of data loss should the program terminate before any consuming goroutines are able to clear the buffer.

Minimizing locking with sharding

As lovely as channels are, as we mentioned in "Efficient Synchronization" on page 201 they don't solve *every* problem. A common example of this is a large, central data structure, such as a cache, that can't be easily decomposed into discrete units of work.[11]

When shared data structures have to be concurrently accessed, it's standard to use a locking mechanism, such as the mutexes provided by the `sync` package, as we do in "Making Your Data Structure Concurrency-Safe" on page 121. For example, we might create a struct that contains a map and an embedded `sync.RWMutex`:

```go
var cache = struct {
    sync.RWMutex
    data map[string]string
}{data: make(map[string]string)}
```

When a routine wants to write to the cache, it would carefully use `cache.Lock` to establish the write lock, and `cache.Unlock` to release the lock when it's done. We might even want to wrap it in a convenience function as follows:

```go
func ThreadSafeWrite(key, value string) {
    cache.Lock()                                // Establish write lock
    cache.data[key] = value
    cache.Unlock()                              // Release write lock
}
```

By design, this restricts write access to whichever routine happens to have the lock. This pattern generally works just fine. However, as we discussed in Chapter 4, as the number of concurrent processes acting on the data increases, the average amount of time that processes spend waiting for locks to be released also increases. You may remember the name for this unfortunate condition: lock contention.

While this might be resolved in some cases by scaling the number of instances, this also increases complexity and latency, as distributed locks need to be established and writes need to establish consistency. An alternative strategy for reducing lock

11 You could probably shoehorn channels into a solution for interacting with a cache, but you might find it difficult to make it simpler than locking.

contention around shared data structures within an instance of a service is *vertical sharding*, in which a large data structure is partitioned into two or more structures, each representing a part of the whole. Using this strategy, only a portion of the overall structure needs to be locked at a time, decreasing overall lock contention.

You may recall that we discussed vertical sharding in some detail in "Sharding" on page 101. If you're unclear on vertical sharding theory or implementation, feel free to take some time to go back and review that section.

Memory Leaks Can...fatal error: runtime: out of memory

Memory leaks are a class of bugs in which memory is not released even after it's no longer needed. These bugs can be quite subtle and often plague languages like C++ in which memory is manually managed. But while garbage collection certainly helps by attempting to reclaim memory occupied by objects that are no longer in use by the program, garbage-collected languages like Go aren't immune to memory leaks. Data structures can still grow unbounded, unresolved goroutines can still accumulate, and even unstopped `time.Ticker` values can get away from you.

In this section we'll review a few common causes of memory leaks particular to Go, and how to resolve them.

Leaking goroutines

I'm not aware of any actual data on the subject,[12] but based purely on my own personal experience, I strongly suspect that goroutines are the single largest source of memory leaks in Go.

Whenever a goroutine is executed, it's initially allocated a small memory stack—2048 bytes—that can be dynamically adjusted up or down as it runs to suit the needs of the process. The precise maximum stack size depends on a lot of things,[13] but it's essentially reflective of the amount of available physical memory.

Normally, when a goroutine returns, its stack is either deallocated or set aside for recycling.[14] Whether by design or by accident, however, not every goroutine actually returns. For example:

```
func leaky() {
    ch := make(chan string)
```

12 If you are, let me know!

13 Dave Cheney wrote an excellent article on this topic called *Why is a Goroutine's stack infinite?* that I recommend you take a look at if you're interested in the dynamics of goroutine memory allocation.

14 There's a very good article by Vincent Blanchon on the subject of goroutine recycling entitled *How Does Go Recycle Goroutines?*

```
        go func() {
            s := <-ch
            fmt.Println("Message:", s)
        }()
    }
```

In the previous example, the `leaky` function creates a channel and executes a goroutine that reads from that channel. The `leaky` function returns without error, but if you look closely you'll see that no values are ever sent to `ch`, so the goroutine will never return and its stack will never be deallocated. There's even collateral damage: because the goroutine references `ch`, that value can't be cleaned up by the garbage collector.

So we now have a bona fide memory leak. If such a function is called regularly the total amount of memory consumed will slowly increase over time until it's completely exhausted.

This is a contrived example, but there are good reasons why a programmer might want to create long-running goroutines, so it's usually quite hard to know whether such a process was created intentionally.

So what do we do about this? Dave Cheney offers some excellent advice here: "You should never start a goroutine without knowing how it will stop.…Every time you use the `go` keyword in your program to launch a goroutine, you must know how, and when, that goroutine will exit. If you don't know the answer, that's a potential memory leak."[15]

This may seem like obvious, even trivial advice, but it's incredibly important. It's all too easy to write functions that leak goroutines, and those leaks can be a pain to identify and find.

Forever ticking tickers

Very often you'll want to add some kind of time dimension to your Go code, to execute it at some point in the future or repeatedly at some interval, for example.

The `time` package provides two useful tools to add such a time dimension to Go code execution: `time.Timer`, which fires at some point in the future, and `time.Ticker`, which fires repeatedly at some specified interval.

However, where `time.Timer` has a finite useful life with a defined start and end, `time.Ticker` has no such limitation. A `time.Ticker` can live forever. Maybe you can see where this is going.

15 Cheney, Dave. "Never Start a Goroutine without Knowing How It Will Stop." dave.cheney.net, 22 Dec. 2016. *https://oreil.ly/VUlrY*.

Both Timers and Tickers use a similar mechanism: each provides a channel that's sent a value whenever it fires. The following example uses both:

```
func timely() {
    timer := time.NewTimer(5 * time.Second)
    ticker := time.NewTicker(1 * time.Second)

    done := make(chan bool)

    go func() {
        for {
            select {
            case <-ticker.C:
                fmt.Println("Tick!")
            case <-done:
                return
            }
        }
    }()

    <-timer.C
    fmt.Println("It's time!")
    close(done)
}
```

The `timely` function executes a goroutine that loops at regular intervals by listening for signals from `ticker`—which occur every second—or from a `done` channel that returns the goroutine. The line `<-timer.C` blocks until the 5-second timer fires, allowing `done` to be closed, triggering the `case <-done` condition and ending the loop.

The `timely` function completes as expected, and the goroutine has a defined return, so you could be forgiven for thinking that everything's fine. There's a particularly sneaky bug here though: running `time.Ticker` values contain an active goroutine that can't be cleaned up. Because we never stopped the timer, `timely` contains a memory leak.

The solution: always be sure to stop your timers. A `defer` works quite nicely for this purpose:

```
func timelyFixed() {
    timer := time.NewTimer(5 * time.Second)
    ticker := time.NewTicker(1 * time.Second)
    defer ticker.Stop()                           // Be sure to stop the ticker!

    done := make(chan bool)

    go func() {
        for {
            select {
            case <-ticker.C:
                fmt.Println("Tick!")
            case <-done:
                return
            }
        }
    }()

    <-timer.C
    fmt.Println("It's time!")
    close(done)
}
```

By calling `ticker.Stop()`, we shut down the underlying `Ticker`, allowing it to be recovered by the garbage collector and preventing a leak.

On Efficiency

In this section, we covered a number of common methods for improving the efficiency of your programs, ranging from using an LRU cache rather than a map to constrain your cache's memory footprint, to approaches for effectively synchronizing your processes, to preventing memory leaks. While these sections might not seem particularly closely connected, they're all important for building programs that scale.

Of course, there are countless other methods that I would have liked to include as well, but wasn't able to given the fundamental limits imposed by time and space.

In the next section, we'll change themes once again to cover some common service architectures and their effects on scalability. These might be a little less focused on Go specifically, but they're critical for a study of scalability, especially in a cloud native context.

Service Architectures

The concept of the *microservice* first appeared in the early 2010s as a refinement and simplification of the earlier service-oriented architecture (SOA) and a response to the *monoliths*—server-side applications contained within a single large executable—that were then the most common architectural model of choice.[16]

At the time, the idea of the microservice architecture—a single application composed of multiple small services, each running in its own process and communicating with lightweight mechanisms—was revolutionary. Unlike monoliths, which require the entire application to be rebuilt and deployed for any change to the system, microservices were independently deployable by fully automated deployment mechanisms. This sounds small, even trivial, but its implications were (and are) vast.

16 Not that they've gone away.

If you ask most programmers to compare monoliths to microservices, most of the answers you get will probably be something about how monoliths are slow, sluggish, and bloated, while microservices are small, agile, and the new hotness. Sweeping generalizations are always wrong, though, so let's take a moment to ask ourselves whether this is true, and whether monoliths might sometimes be the right choice.

We will begin by defining what we mean when we talk about monoliths and microservices.

The Monolith System Architecture

In a *monolith architecture*, all of the functionally distinguishable aspects of a service are coupled together in one place. A common example is a web application whose user interface, data layer, and business logic are all intermingled, often on a single server.

Traditionally, enterprise applications have been built in three main parts, as illustrated in Figure 7-3: a client-side interface running on the user's machine, a relational database where all of the application's data lives, and a server-side application that handles all user input, executes all business logic, and reads and writes data to the database.

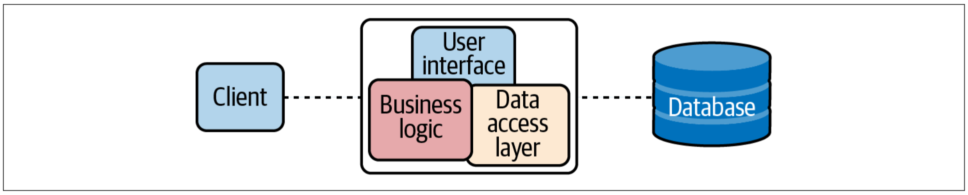


Figure 7-3. In a monolith architecture, all of the functionally distinguishable aspects of a service are coupled together in one place

At the time, this pattern made sense. All the business logic ran in a single process, making development easier, and you could even scale by running more monoliths behind a load balancer, usually using sticky sessions to maintain server affinity. Things were *perfectly fine*, and for many years this was by far the most common way of building web applications.

Even today, for relatively small or simple applications (for some definition of "small" and "simple") this works perfectly well (though I still strongly recommend statelessness over server affinity).

However, as the number of features and general complexity of a monolith increases, difficulties start to arise:

- Monoliths are usually deployed as a single artifact, so making even a small change generally requires a new version of the entire monolith to be built, tested, and deployed.
- Despite even the best of intentions and efforts, monolith code tends to decrease in modularity over time, making it harder to make changes in one part of the service without affecting another part in unexpected ways.
- Scaling the application means creating replicas of the entire application, not just the parts that need it.

The larger and more complex the monolith gets, the more pronounced these effects tend to become. By the early- to mid-2000s, these issues were well known, leading frustrated programmers to experiment with breaking their big, complex services into smaller, independently deployable and scalable components. By 2012, this pattern even had a name: microservices architecture.

The Microservices System Architecture

The defining characteristic of a *microservices architecture* is a service whose functional components have been divided into a set of discrete sub-services that can be independently built, tested, deployed, and scaled.

This is illustrated in Figure 7-4, in which a user interface service—perhaps an HTML-serving web application or a public API—interacts with clients, but rather than handling the business logic locally, it makes secondary requests of one or more component services to handle some specific functionality. Those services might in turn even make further requests of yet more services.

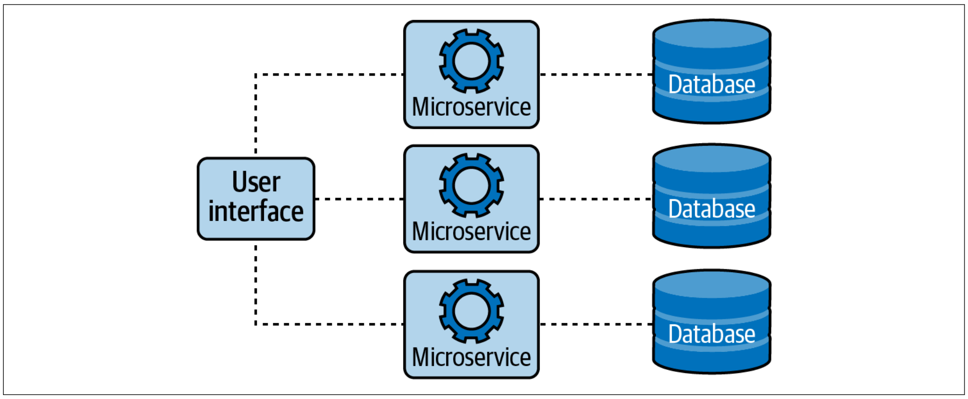


Figure 7-4. In a microservices architecture, functional components are divided into discrete subservices

While the microservices architecture has a number of advantages over the monolith, there are significant costs to consider. On one hand, microservices provide some significant benefits:

- A clearly-defined separation of concerns supports and reinforces modularity, which can be very useful for larger or multiple teams.
- Microservices should be independently deployable, making them easier to manage and making it possible to isolate errors and failures.
- In a microservices system, it's possible for different services to use the technology—language, development framework, data storage, etc—that's most appropriate to its function.

These benefits shouldn't be underestimated: the increased modularity and functional isolation of microservices tends to produce components that are themselves generally far more maintainable than a monolith with the same functionality. The resulting system isn't just easier to deploy and manage, but easier to understand, reason about, and extend for a larger number of programmers and teams.

> Mixing different technologies may sound appealing in theory, but use restraint. Each adds new requirements for tooling and expertise. The pros and cons of adopting a new technology—any new technology[17]—should always be carefully considered.

The discrete nature of microservices makes them far easier to maintain, deploy, and scale than monoliths. However, while these are real benefits that can pay real dividends, there are some downsides as well:

- The distributed nature of microservices makes them subject to the Fallacies of Distributed Computing (see Chapter 4), which makes them significantly harder to program and debug.
- Sharing any kind of state between your services can often be extremely difficult.
- Deploying and managing multiple services can be quite complex and tends to demand a high level of operational maturity.

So given these, which do you choose? The relative simplicity of the monolith, or the flexibility and scalability of microservices? You might have noticed that most of the benefits of microservices pay off as the application gets larger or the number of teams

17 Yes, even Go.

working on it increases. For this reason many authors advocate starting with a monolith and decomposing it later.

On a personal note, I'll mention that I've never seen any organization successfully break apart a large monolith, but I've seen many try. That's not to say it's impossible, just that it's hard. I can't tell you whether you should start your system as microservices, or with a monolith and break it up later. I'd certainly get a lot of angry emails if I tried. But please, whatever you do, stay stateless.

Serverless Architectures

Serverless computing is a pretty popular topic in web application architecture, and a lot of (digital) ink has been spilled about it. Much of this hype has been driven by the major cloud providers, which have invested heavily in serverlessness, but not all of it.

But what is serverless computing, really?

Well, as is often the case, it depends on who you ask. For the purposes of this book, however, we're defining it as a form of utility computing in which some server-side logic, written by a programmer, is transparently executed in a managed ephemeral environment in response to some predefined trigger. This is also sometimes referred to as "functions as a service," or "FaaS." All of the major cloud providers offer FaaS implementations, such as AWS's Lambda or GCP's Cloud Functions.

Such functions are quite flexible and can be usefully incorporated into many architectures. In fact, as we'll discuss shortly, entire *serverless architectures* can even be built that don't use traditional services at all, but are instead built entirely from FaaS resources and third-party managed services.

Be Suspicious of Hype

I may sound like a grizzled old dinosaur here, but I've learned to be wary of new technologies that nobody really understands claiming to solve all of our problems.

According to the research and advisory firm Gartner, which specializes in studying IT and technology trends, serverless infrastructure is hovering at or near the "Peak of Inflated Expectations"[18] of its "hype cycle". This is eventually, but inevitably, followed by the "Trough of Disillusionment."

In time, people start to figure out what the technology is really useful for (not *everything*) and when to use it (not *always*), and it enters the "Slope of Enlightenment" and

18 Bowers, Daniel, et al. "Hype Cycle for Compute Infrastructure, 2019." *Gartner*, Gartner Research, 26 July 2019, *https://oreil.ly/3gkJh*.

"Plateau of Productivity." I've learned the hard way that it's usually best to wait until a technology has entered these two later phases before investing heavily in its use.

That being said: serverless computing *is* intriguing, and it *does* seem appropriate for some use-cases.

The pros and cons of serverlessness

As with any other architectural decision, the choice to go with a partially or entirely serverless architecture should be carefully weighed against all available options. While serverlessness provides some clear benefits—some obvious (no servers to manage!), others less so (cost and energy savings)—it's very different from traditional architectures, and carries its own set of downsides.

That being said, let's start weighing. Let's start with the advantages:

Operational management
: Perhaps the most obvious benefit of serverless architectures is that there's considerably less operational overhead.[19] There are no servers to provision and maintain, no licenses to buy, and no software to install.

Scalability
: When using serverless functions, it's the provider—not the user—who's responsible for scaling capacity to meet demand. As such, the implementor can spend less time and effort considering and implementing scaling rules.

Reduced costs
: FaaS providers typically use a "pay-as-you-go" model, charging only for the time and memory allocated when the function is run. This can be considerably more cost-effective than deploying traditional services to (likely underutilized) servers.

Productivity
: In a FaaS model, the unit of work is an event-driven function. This model tends to encourage a "function first" mindset, resulting in code that's often simpler, more readable, and easier to test.

It's not all roses, though. There are very some real downsides to serverless architectures that need to be taken into consideration as well:

Startup latency
: When a function is first called, it has to be "spun up" by the cloud provider. This typically takes less than a second, but in some cases can add 10 or more seconds to the initial requests. This is known as the *cold start* delay. What's more, if the

19 It's *right in the name!*

function isn't called for several minutes—the exact time varies between providers—it's "spun down" by the provider so that it has to endure another cold start when it's called again. This usually isn't a problem if your function doesn't have enough idle time to get spun down, but can be a significant issue if your load is particularly "bursty."

Observability
: While most of the cloud vendors provide some basic monitoring for their FaaS offerings, it's usually quite rudimentary. While third-party providers have been working to fill the void, the quality and quantity of data available from your ephemeral functions is often less than desired.

Testing
: While unit testing tends to be pretty straightforward for serverless functions, integration testing is quite hard. It's often difficult or impossible to simulate the serverless environment, and mocks are approximations at best.

Cost
: Although the "pay-as-you-go" model can be considerably cheaper when demand is lower, there is a point at which this is no longer true. In fact, very high levels of load can grow to be quite expensive.

Clearly, there's quite a lot to consider—on both sides—and while there *is* a great deal of hype around serverless at the moment, to some degree I think it's merited. However, while serverlessness promises (and largely delivers) scalability and reduced costs, it does have quite a few gotchas, including, but not limited to, testing and debugging challenges. Not to mention the increased burden on operations around observability![20]

Finally, as we'll see in the next section, serverless architectures also require quite a lot more up-front planning than traditional architectures. While some people might call this a positive feature, it can add significant complexity.

Serverless services

As mentioned previously, functions as a service (FaaS) are flexible enough to serve as the foundation of entire serverless architectures that don't use traditional services at all, but are instead built entirely from FaaS resources and third-party managed services.

20 Sorry, there's no such thing as NoOps.

Let's take, as an example, the familiar three-tier system in which a client issues a request to a service, which in turn interacts with a database. A good example is the key-value store we started in Chapter 5, whose (admittedly primitive) monolithic architecture might look something like what's shown in Figure 7-5.

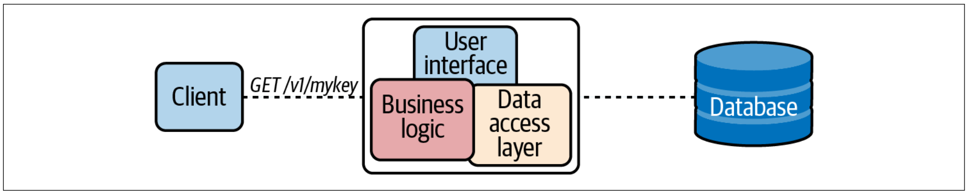


Figure 7-5. The monolithic architecture of our primitive key/value store

To convert this monolith into a serverless architecture, we'll need to use an *API gateway*: a managed service that's configured to expose specific HTTP endpoints and to direct requests to each endpoint to a specific resource—typically a FaaS functions—that handles requests and issue responses. Using this architecture, our key/value store might look something like what's shown in Figure 7-6.

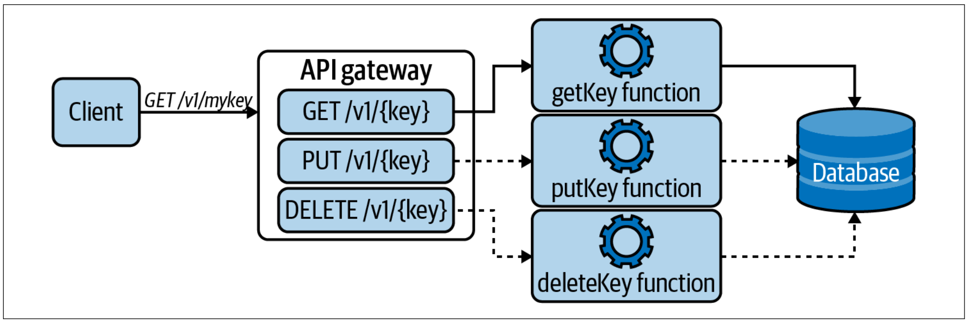


Figure 7-6. An API gateway routes HTTP calls to serverless handler functions

In this example, we've replaced the monolith with an API gateway that supports three endpoints: `GET /v1/{key}`, `PUT /v1/{key}`, and `DELETE /v1/{key}` (the `{key}` component indicates that this path will match any string, and refer to the value as `key`).

The API gateway is configured so that requests to each of its three endpoints are directed to a different handler function—`getKey`, `putKey`, and `deleteKey`, respectively—which performs all of the logic for handling that request and interacting with the backing database.

Granted, this is an incredibly simple application and doesn't account for things like authentication (which can be provided by a number of excellent third-party services like Auth0 or Okta), but some things are immediately evident.

First, there are a greater number of moving parts that you have to get your head around, which necessitates quite a bit more up-front planning and testing. For example, what happens if there's an error in a handler function? What happens to the request? Does it get forwarded to some other destination, or is it perhaps sent to a dead-letter queue for further processing?

Do not underestimate the significance of this increase in complexity! Replacing in-process interactions with distributed, fully managed components tends to introduce a variety of problems and failure cases that simply don't exist in the former. You may well have turned a relatively simple problem into an enormously complex one. Complexity kills; simplicity scales.

Second, with all of these different components, there's a need for more sophisticated distributed monitoring than you'd need with a monolith or small microservices system. Due to the fact that FaaS relies heavily on the cloud provider, this may be challenging or, at least, awkward.

Finally, the ephemeral nature of FaaS means that ALL state, even short-lived optimizations like caches, has to be externalized to a database, an external cache (like Redis), or network file/object store (like S3). Again, this can be argued to be a Good Thing, but it does add to up-front complexity.

Summary

This was a very difficult chapter to write, not because there isn't much to say, but because scalability is such a huge topic with so many different things I could have drilled down into. Every one of these battled in my brain for weeks.

I even ended up throwing away some perfectly good architecture content that, in retrospect, simply wasn't appropriate for this book. Fortunately, I was able to salvage a whole other chunk of work about messaging that ended up getting moved into Chapter 8. I think it's happier there anyway.

In those weeks, I spent a lot of time thinking about what scalability really is, and about the role that efficiency plays in it. Ultimately, I think that the decision to spend so much time on programmatic—rather than infrastructural—solutions to scaling problems was the right one.

All told, I think the end result is a good one. We certainly covered a lot of ground:

- We reviewed the different axes of scaling, and how scaling out is often the best long-term strategy.
- We discussed state and statelessness, and why application state is essentially "anti-scalability."

- We learned a few strategies for efficient in-memory caching and for avoiding memory leaks.
- We compared and contrasted monolithic, microservice, and serverless architectures.

That's quite a lot, and although I wish I'd been able to drill down in some more detail, I'm pleased to have been able to touch on the things I did.

CHAPTER 8
Loose Coupling

We build our computers the way we build our cities—over time, without a plan, on top of ruins.[1]

—Ellen Ullman, *The Dumbing-down of Programming (May 1998)*

Coupling is one of those fascinating topics that seem straightforward in theory but are actually quite challenging in practice. As we'll discuss, there are lots of ways in which coupling can be introduced in a system, which means it's also a *big* subject. As you might imagine, this chapter is an ambitious one, and we cover a lot of ground.

First, we'll introduce the subject, diving more deeply into the concept of "coupling," and discussing the relative merits of "loose" versus "tight" coupling. We'll present some of the most common coupling mechanisms, and how some kinds of tight coupling can lead to the dreaded "distributed monolith."

Next, we'll talk about inter-service communications, and how fragile exchange protocols are a very common way of introducing tight coupling to distributed systems. We'll cover some of the common protocols in use today to minimize the degree of coupling between two services.

In the third part, we'll change direction for a bit, away from distributed systems and into the implementations of the services themselves. We'll talk about services as code artifacts, subject to coupling resulting from mingling implementations and violating separation of concerns, and present the use of plug-ins as a way to dynamically add implementations.

Finally, we'll close with a discussion of hexagonal architecture, an architectural pattern that makes loose coupling the central pillar of its design philosophy.

1 Ullman, Ellen. "The Dumbing-down of Programming." *Salon*, 12 May 1998. *https://oreil.ly/Eib3K*.

Throughout the chapter, we'll do our best to balance theory, architecture, and implementation. Most of the chapter will be spent on the fun stuff: discussing a variety of different strategies for managing coupling, particularly (but not exclusively) in the distributed context, and demonstrating by extending our example key/value store.

Tight Coupling

"Coupling" is a somewhat romantic term describing the degree of direct knowledge between components. For example, a client that sends requests to a service is by definition coupled to that service. The degree of that coupling can vary considerably, however, falling anywhere between two extremes.

"Tightly coupled" components have a great deal of knowledge about another component. Perhaps both require the same version of a shared library to communicate, or maybe the client needs an understanding of the server's architecture or database schema. It's easy to build tightly coupled systems when optimizing for the short term, but they have a huge downside: the more tightly coupled two components are, the more likely that a change to one component will necessitate corresponding changes to the other. As a result, tightly coupled systems lose many of the benefits of a microservice architecture.

In contrast, "loosely coupled" components have minimal direct knowledge of one another. They're relatively independent, typically interacting via a change-robust abstraction. Systems designed for loose coupling require more up-front planning, but they can be more freely upgraded, redeployed, or even entirely rewritten without greatly affecting the systems that depend on them.

Put simply, if you want to know how tightly coupled your system is, ask how many and what kind of changes can be made to one component without adversely affecting another.

Some amount of coupling isn't necessarily a bad thing, especially early in a system's development. It can be temping to over-abstract and over-complicate, but premature optimization is *still* the root of all evil.

Coupling in Different Computing Contexts

The term "coupling" in the computing context predates microservices and service-oriented architecture by quite a bit, and has been used for many years to describe the degree of knowledge that one component has about another.

In programming, code can be tightly coupled when a dependent class directly references a concrete implementation instead of an abstraction (such as an interface; see

"Interfaces" on page 59). In Go this might be a function that requires an `os.File` when an `io.Reader` would do.

Multiprocessor systems that communicate by sharing memory can be said to be tightly coupled. In a loosely coupled system, components are connected through a MTS (Message Transfer System) (see "Efficient Synchronization" on page 201 for a refresher on how Go solves this problem with channels).

It's important to note that there might, on occasion, be good reason to tightly couple certain components. Eliminating abstractions and other intermediate layers can reduce overhead, which can be a useful optimization if speed is a critical system requirement.

Since this book is largely about distributed architectures, we'll focus on coupling between services that communicate across a network, but keep in mind that there are other ways that software can be tightly coupled to resources in its environment.

Tight Coupling Takes Many Forms

There's no limit to the ways components in a distributed system can find themselves tightly coupled. However, while these all share one fundamental flaw—they all depend on some property of another component that they wrongly assume won't change—most can be grouped into a few broad classes according to the resource that they're coupled to.

Fragile exchange protocols

Remember SOAP (simple object access protocol)? Statistically speaking, probably not.[2] SOAP was a messaging protocol developed in the late 1990s that was designed for extensibility and implementation neutrality. SOAP services provided a *contract* that clients could follow to format their requests.[3] The concept of the contract was something of a breakthrough at the time, but SOAP's implementation was exceedingly fragile: if the contract changed in any way, the clients had to be updated along with it. This requirement meant that SOAP clients were tightly coupled to their services.

It didn't take long for people to realize that this was a problem, and SOAP quickly lost its shine. It's since been largely replaced by REST, which, while a considerable improvement, can often introduce its own tight coupling. In 2016, Google released

2 Get off my lawn.

3 In XML, no less. We didn't know any better at the time.

gRPC (gRPC Remote Procedure Calls[4]), an open source framework with a number of useful features, including, importantly, allowing loose coupling between components.

We'll discuss some of these more contemporary options in "Communications Between Services" on page 224, where we'll see how to use Go's `net/http` package to build a REST/HTTP client and extend our key/value store with a gRPC frontend.

Shared dependencies

In 2016, Facebook's Ben Christensen gave a talk at the Microservices Practitioner Summit where he spoke about another increasingly common mechanism for tightly coupling distributed services, introducing the term "distributed monolith" in the process.

Ben described an anti-pattern in which services were *required* to use specific libraries—and versions of libraries—in order to launch and interact with one another. Such systems find themselves saddled with a fleet-wide dependency, such that upgrading these shared libraries can force all services to have to upgrade in lockstep. This shared dependency has tightly coupled all of the services in the fleet.

Distributed Monoliths

In Chapter 7 we made the case that monoliths, at least for complex systems with multiple distinct functions, are (generally) less desirable, and microservices are (generally) the way to go.[5] Of course, that's easier said than done, in large part because it's so easy to accidentally create a *distributed monolith*: a microservice-based system containing tightly coupled services.

In a distributed monolith, even small changes to one service can necessitate changes to others, often triggering unintended consequences. Services often can't be deployed independently, so deployments have to be carefully orchestrated, and errors in one component can send faults rippling through the entire system. Rollbacks are functionally impossible.

In other words, a distributed monolith is a "worst of all worlds" system that pairs the management and complexity overhead of multiple services with the dependencies and entanglements of a monolith, losing many of the benefits of microservices in the process. Avoid at all costs.

4 At Google, even the acronyms are recursive.

5 This is actually a pretty nuanced discussion. See "Service Architectures" on page 209.

Shared point-in-time

Often systems are designed in such a way that clients expect an immediate response from services. Systems using this *request-response messaging* pattern implicitly assume that a service is present and ready to promptly respond. But if it's not, the request will fail. It can be said that they're *coupled in time*.

Coupling in time isn't necessarily bad practice, though. It might even be preferable, particularly when there's a human waiting for a timely response. We even detail how to construct such a client in the section "Request-Response Messaging" on page 224.

But if the response isn't necessarily time-constrained, then a safer approach may be to send messages to an intermediate queue that recipients can retrieve from when they're ready, a messaging pattern commonly referred to as *publish-subscribe messaging* ("pub-sub" for short).

Fixed addresses

It's the nature of microservices that they need to talk to one another. But to do that, they first have to find each other. This process of locating services on a network is called *service discovery*.

Traditionally, services lived at relatively fixed, well-known network locations that could be discovered by referencing some centralized registry. Initially this took the form of manually maintained `hosts.txt` files, but as networks scaled up so did the adoption of DNS and URLs.

Traditional DNS works well for long-lived services whose locations on the network rarely change, but the increased popularity of ephemeral, microservice-based applications has ushered in a world in which the lifespans of service instances is often measurable in seconds or minutes rather than months or years. In such dynamic environments, URLs and traditional DNS become just another form of tight coupling.

This need for dynamic, fluid service discovery has driven the adoption of entirely new strategies like the *service mesh*, a dedicated layer for facilitating service-to-service communications between resources in a distributed system.

> Unfortunately, we won't be able to cover the fascinating and fast-developing topics of service discovery or service meshes in this book. But the service mesh field is rich, with a number of mature open source projects with active communities—such as Envoy, Linkerd, and Istio—and even commercial offerings like Hashicorp's Consul.

Communications Between Services

Communication and message passing is a critical function of distributed systems, and all distributed systems depend on some form of messaging to receive instructions and directions, exchange information, and provide results and updates. Of course, a message is useless if the recipient can't understand it.

In order for services to communicate, they must first establish an implicit or explicit *contract* that defines how messages will be structured. While such a contract is necessary, it also effectively couples the components that depend on it.

It's actually very easy to introduce tight coupling in this way, the degree of which is reflected in the protocol's ability to change safely. Does it allow backward- and forward-compatible changes, like protocol buffers and gRPC, or do minor changes to the contract effectively break communications, as is the case with SOAP?

Of course, the data exchange protocol and its contract isn't the only variable in inter-service communications. There are, in fact, two broad classes of messaging patterns:

Request-response (synchronous)
: A two-way message exchange in which a requester (the client) issues a request of a receiver (the service) and waits for a response. A textbook example is HTML.

Publish-subscribe (asynchronous)
: A one-way message exchange in which a requester (the publisher) issues a message to an event bus or message exchange, rather than directly to a specific receiver. Messages can be retrieved asynchronously and acted upon by one or more services (subscribers).

Each of these patterns has a variety of implementations and particular use-cases, each with their own pros and cons. While we won't be able to cover every possible nuance, we'll do our best to provide a usable survey and some direction about how they may be implemented in Go.

Request-Response Messaging

As its name suggests, systems using a *request-response*, or *synchronous*, messaging pattern communicate using a series of coordinated requests and responses, in which a requester (or client) submits a request to a receiver (or service) and waits until the receiver responds (hopefully) with the requested the data or service (see Figure 8-1).

The most obvious example of this pattern might be HTTP, which is so ubiquitous and well-established that it's been extended beyond its original purpose, and now underlies common messaging protocols like REST and GraphQL.

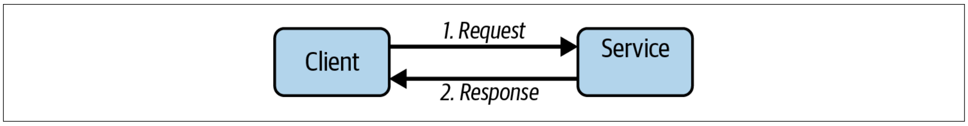


Figure 8-1. Systems using a request-response messaging pattern communicate using a series of coordinated requests and responses

The request-response pattern has the advantages of being relatively easy to reason about and straightforward to implement, and has long been considered the default messaging pattern, particularly for public-facing services. However, it's also "point-to-point," involving exactly one requester and receiver, and requires the requesting process to pause until it receives a response.

Together, these properties make the request-response pattern a good choice for straightforward exchanges between two endpoints where a response can be expected in a reasonably short amount of time, but less than ideal when a message has to be sent to multiple receivers or when a response might take longer than a requester might want to wait.

Common Request-Response Implementations

Over the years, a multitude of bespoke request-response protocols have been developed for any number of purposes. Over time, this has largely settled down, giving way to three major implementations.

REST
: You're likely already very familiar with REST, which we discussed in some detail in "Building an HTTP Server with net/http" on page 112. REST has some things going for it. It's human-readable and easy to implement, making it a good choice for outward-facing services (which is why we chose it in Chapter 5). We'll discuss a little more in "Issuing HTTP Requests with net/http" on page 226.

Remote procedure calls (RPC)
: Remote procedure call (RPC) frameworks allow programs to execute procedures in a different address space, often on another computer. Go provides a standard Go-specific RPC implementation in the form of `net/rpc`. There are also two big language-agnostic RPC players: Apache Thrift and gRPC. While similar in design and usage goals, gRPC seems to have taken the lead with respect to adoption and community support. We'll discuss gRPC in much more detail in "Remote Procedure Calls with gRPC" on page 230.

GraphQL
: A relative newcomer on the scene, GraphQL is a query and manipulation language generally considered an alternative to REST, and is particularly powerful when working with complex datasets. We don't discuss GraphQL in much detail

in this book, but I encourage you to look into it the next time you're designing an outward-facing API.

Issuing HTTP Requests with net/http

HTTP is perhaps the most common request-response protocol, particularly for public-facing services, underlying popular API formats like REST and GraphQL. If you're interacting with an HTTP service, you'll need some way to programmatically issue requests to the service and retrieve the response.

Fortunately, the Go standard library comes with excellent HTTP client and server implementations in the form of the `net/http` package. You may remember `net/http` from "Building an HTTP Server with net/http" on page 112, where we used it to build the first iteration of our key/value store.

The `net/http` includes, among other things, convenience functions for GET, HEAD, and POST methods. The signatures for the first of these, `http.Get` and `http.Head`, are shown in the following:

```go
// Get issues a GET to the specified URL
func Get(url string) (*http.Response, error)

// Head issues a HEAD to the specified URL
func Head(url string) (*http.Response, error)
```

The previous functions are very straightforward and are both used similarly: each accepts a `string` that represents the URL of interest, and each returns an `error` value and a pointer to an `http.Response` struct.

The `http.Response` struct is particularly useful because it contains all kinds of useful information about the service's response to our request, including the returned status code and the response body.

A small selection of the `http.Response` struct is in the following:

```go
type Response struct {
    Status     string       // e.g. "200 OK"
    StatusCode int          // e.g. 200

    // Header maps header keys to values.
    Header Header

    // Body represents the response body.
    Body io.ReadCloser

    // ContentLength records the length of the associated content. The
    // value -1 indicates that the length is unknown.
    ContentLength int64

    // Request is the request that was sent to obtain this Response.
```

```
        Request *Request
    }
```

There are some useful things in there! Of particular interest is the `Body` field, which provides access to the HTTP response body. It's a `ReadCloser` interface, which tells us two things: that the response body is streamed on demand as it's read, and that it has a `Close` method that we're expected to call.

In the following, we demonstrate several things: how to use the `Get` convenience function, how to close the response body, and how to use `io.ReadAll` to read the *entire* response body as a string (if you're into that kind of thing):

```
package main

import (
    "fmt"
    "io"
    "net/http"
)

func main() {
    resp, err := http.Get("http://example.com")     // Send an HTTP GET
    if err != nil {
        panic(err)
    }
    defer resp.Body.Close()                         // Close your response!

    body, err := io.ReadAll(resp.Body)          // Read body as []byte
    if err != nil {
        panic(err)
    }

    fmt.Println(string(body))
}
```

In the last example, we use the `http.Get` function to issue a GET to the URL `http://example.com`, which returns a pointer to a `http.Response` struct and an `error` value.

As we mentioned previously, access to the HTTP response body is provided via the `resp.Body` variable, which implements `io.ReadCloser`. Note how we `defer` the call `resp.Body.Close()`. This is very important: failing to close your response body can sometimes lead to some unfortunate memory leaks.

Because `Body` implements `io.Reader`, we have many different standard means to retrieve its data. In this case we use the very reliable `io.ReadAll`, which conveniently returns the entire response body as a `[]byte` slice, which we simply print.

Always remember to use `Close()` to close your response body! Not doing so can lead to some unfortunate memory leaks.

We've already seen the `Get` and `Head` functions, but how do we issue POSTs? Fortunately, similar convenience functions exist for them too. Two, in fact: `http.Post` and `http.PostForm`. The signatures for each of these are shown in the following:

```
// Post issues a POST to the specified URL
func Post(url, contentType string, body io.Reader) (*Response, error)

// PostForm issues a POST to the specified URL, with data's keys
// and values URL-encoded as the request body
func PostForm(url string, data url.Values) (*Response, error)
```

The first of these, `Post`, expects an `io.Reader` that provides the body—such as a file of a JSON object—of the post. We demonstrate how to upload JSON text in a POST in the following code:

```
package main

import (
    "fmt"
    "io"
    "net/http"
    "strings"
)

const json = `{ "name":"Matt", "age":44 }`      // This is our JSON

func main() {
    in := strings.NewReader(json)               // Wrap JSON with an io.Reader

    // Issue HTTP POST, declaring our content-type as "text/json"
    resp, err := http.Post("http://example.com/upload", "text/json", in)
    if err != nil {
        panic(err)
    }
    defer resp.Body.Close()                     // Close your response!

    message, err := io.ReadAll(resp.Body)
    if err != nil {
        panic(err)
    }

    fmt.Printf(string(message))
}
```

A Possible Pitfall of Convenience Functions

We've been referring to the `Get`, `Head`, `Post`, and `PostForm` functions as "convenience functions," but what does that mean?

It turns out that, under the hood, each is actually calling a method on a default `*http.Client` value, a concurrency-safe type that Go uses to manage the internals of communicating over HTTP.

The code for the `Get` convenience function, for example, is actually a call to the default client's `http.Client.Get` method:

```
func Get(url string) (resp *Response, err error) {
    return DefaultClient.Get(url)
}
```

As you can see, when you use `http.Get`, you're actually using `http.DefaultClient`. Because `http.Client` is concurrency safe, it's possible to have only one of these, predefined as a package variable.

The source code for the creation of `DefaultClient` itself is somewhat plain, creating a zero-value `http.Client`:

```
var DefaultClient = &Client{}
```

Generally, this is perfectly fine. However, there's a potential issue here, and it involves timeouts. The `http.Client` methods are capable of asserting timeouts that terminate long-running requests. This is super useful. Unfortunately, the default timeout value is 0, which Go interprets as "no timeout."

Okay, so Go's default HTTP client will never time out. Is that a problem? *Usually* not, but what if it connects to a server that doesn't respond and doesn't close the connection? The result would be an especially nasty and nondeterministic memory leak.

But how do we fix this? Well, as it turns out, `http.Client` *does* support timeouts; we just have to enable that functionality by creating a custom `Client` and setting a timeout:

```
var client = &http.Client{
    Timeout: time.Second * 10,
}
response, err := client.Get(url)
```

Take a look at the net/http package documentation for more information about `http.Client` and its available settings.

Remote Procedure Calls with gRPC

gRPC is an efficient, polyglot data exchange framework that was originally developed by Google as the successor to *Stubby*, a general-purpose RPC framework that had been in use internally at Google for over a decade. It was open sourced in 2015 under the name gRPC, and taken over by the Cloud Native Computing Foundation in 2017.

Unlike REST, which is essentially a set of unenforced best practices, gRPC is a fully-featured data exchange framework, which, like other RPC frameworks such as SOAP, Apache Thrift, Java RMI, and CORBA (to name a few) allows a client to execute specific methods implemented on different systems as if they were local functions.

This approach has a number of advantages over REST, including but not limited to:

Conciseness
: Its messages are more compact, consuming less network I/O.

Speed
: Its binary exchange format is much faster to marshal and unmarshal.

Strong-typing
: It's natively strongly typed, eliminating a lot of boilerplate and removing a common source of errors.

Feature-rich
: It has a number of built-in features like authentication, encryption, timeout, and compression (to name a few) that you would otherwise have to implement yourself.

That's not to say that gRPC is always the best choice. Compared to REST:

Contract-driven
: gRPC's contracts make it less suitable for external-facing services.

Binary format
: gRPC data isn't human-readable, making it harder to inspect and debug.

> gRPC is a very large and rich subject that this modest section can't fully do justice to. If you're interested in learning more I recommend the official "Introduction to gRPC" and the excellent *gRPC: Up and Running* by Kasun Indrasiri and Danesh Kuruppu (O'Reilly Media).

Interface definition with protocol buffers

As is the case with most RPC frameworks, gRPC requires you to define a *service interface*. By default, gRPC uses *protocol buffers* for this purpose, though it's possible to use an alternative Interface Definition Language (IDL) like JSON if you want.

To define a service interface, the author uses the protocol buffers schema to describe the service methods that can be called remotely by a client in a `.proto` file. This is then *compiled* into a language-specific interface (Go code, in our case).

As illustrated in Figure 8-2, gRPC servers implement the resulting source code to handle client calls, while the client has a stub that provides the same methods as the server.

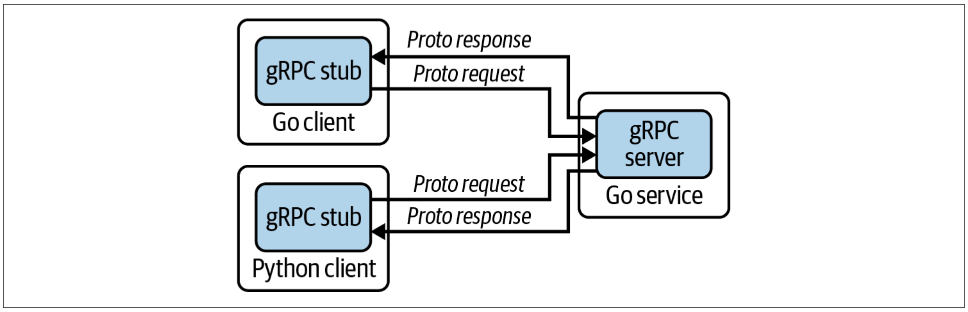


Figure 8-2. By default, gRPC uses protocol buffers as both its Interface Definition Language and its underlying message interchange format; servers and clients can be written in any supported language

Yes, this seems very hand-wavey and abstract right now. Keep reading for some more details!

Installing the protocol compiler

Before we proceed, we'll first need to install the protocol buffer compiler, `protoc`, and the Go protocol buffers plug-in. We'll use these to compile `.proto` files into Go service interface code.

1. If you're using Linux or MacOS, the simplest and easiest way to install `protoc` is to use a package manager. To install it on a Debian-flavored Linux you can use `apt` or `apt-get`:

   ```
   $ apt install -y protobuf-compiler
   $ protoc --version
   ```

The easiest way to install `protoc` on MacOS is to use Homebrew:

```
$ brew install protobuf
$ protoc --version
```

2. Run the following command to install the Go protocol buffers plug-in:

```
$ go install google.golang.org/protobuf/cmd/protoc-gen-go
```

The compiler plug-in `protoc-gen-go` will be installed in `$GOBIN`, defaulting to `$GOPATH/bin`. It must be in your `$PATH` for `protoc` to find it.

> This book uses protocol buffers version 3. Be sure to check your version of `protoc` after installation to make sure that it's version 3 or higher.

If you're using another OS, your chosen package manager has an old version, or if you just want to make sure you have the latest and greatest, you can find the instructions for installing the precompiled binaries on gRPC's Protocol Buffer Compiler Installation page.

The message definition structure

Protocol buffers are a language-neutral mechanism for serializing structured data. You can think of it as a binary version of XML.[6] Protocol buffer data is structured as *messages*, where each message is a small record of information containing a series of name-value pairs called *fields*.

The first step when working with protocol buffers is to define the message structure by defining it in a `.proto` file. A basic example is presented in the following:

Example 8-1. An example `.proto` file. The `message` definitions define remote procedure payloads.

```
syntax = "proto3";

option go_package = "github.com/cloud-native-go/ch08/point";

// Point represents a labeled position on a 2-dimensional surface
message Point {
  int32 x = 1;
  int32 y = 2;
  string label = 3;
```

6 If you're into that kind of thing.

```
}

// Line contains start and end Points
message Line {
  Point start = 1;
  Point end = 2;
  string label = 3;
}

// Polyline contains any number (including zero) of Points
message Polyline {
  repeated Point point = 1;
  string label = 2;
}
```

You may have noticed that the protocol buffer syntax is reminiscent of C/C++, complete with its semicolon and commenting syntax.

The first line of the file specifies that you're using `proto3` syntax: if you don't do this, the protocol buffer compiler will assume you are using `proto2`. This must be the first nonempty, noncomment line of the file.

The second line uses the `option` keyword to specify the full import path of the Go package that will contain the generated code.

Finally, we have three `message` definitions, which describe the structure of the payload messages. In this example, we have three messages of increasing complexity:

- `Point`, which contains `x` and `y` integer values, and a `label` string
- `Line`, which contains exactly two `Point` values
- `Polyline`, which uses the `repeated` keyword to indicate that it can contain any number of `Point` values

Each `message` contains zero or more fields that have a name and a type. Note that each field in a message definition has a *field number* that is unique for that message type. These are used to identify fields in the message binary format, and should not be changed once your message type is in use.

If this raises a "tight coupling" red flag in your mind, you get a gold star for paying attention. For this reason, protocol buffers provide explicit support for updating message types, including marking a field as reserved so that it can't be accidentally reused.

This example is incredibly simple, but don't let that fool you: protocol buffers are capable of some very sophisticated encodings. See the Protocol Buffers Language Guide for more information.

The key-value message structure

So how do we make use of protocol buffers and gRPC to extend the example key-value store that we started in Chapter 5?

Let's say that we want to implement gRPC equivalents to the `Get`, `Put`, and `Delete` functions we are already exposing via RESTful methods. The message formats for that might look something like the following `.proto` file:

Example 8-2. `keyvalue.proto`—the messages that will be passed to and from our key-value service procedures

```
syntax = "proto3";

option go_package = "github.com/cloud-native-go/ch08/keyvalue";

// GetRequest represents a request to the key-value store for the
// value associated with a particular key
message GetRequest {
  string key = 1;
}

// GetResponse represents a response from the key-value store for a
// particular value
message GetResponse {
  string value = 1;
}

// PutRequest represents a request to the key-value store for the
// value associated with a particular key
message PutRequest {
  string key = 1;
  string value = 2;
}

// PutResponse represents a response from the key-value store for a
// Put action.
message PutResponse {}

// DeleteRequest represents a request to the key-value store to delete
// the record associated with a key
message DeleteRequest {
  string key = 1;
}

// DeleteResponse represents a response from the key-value store for a
// Delete action.
message DeleteResponse {}
```

Don't let the names of the message definitions confuse you: they represent *messages* (nouns) that will be passed to and from functions (verbs) that we'll define in the next section.

In the *.proto* file, which we'll call *keyvalue.proto*, we have three `Request` message definitions describing messages that will be sent from the client to the server, and three `Response` message definitions describing the server's response messages.

You may have noticed that we don't include `error` or `status` values in the message response definitions. As you'll see in "Implementing the gRPC client" on page 239, these are unnecessary because they're included in the return values of the gRPC client functions.

Defining our service methods

So now that we've completed our message definitions, we'll need to describe the methods that'll use them.

To do that, we extend our *keyvalue.proto* file, using the `rpc` keyword to define our service interfaces. Compiling the modified *.proto* file will generate Go code that includes the service interface code and client stubs.

Example 8-3. `keyvalue.proto`—the procedures for our key-value service.

```
service KeyValue {
  rpc Get(GetRequest) returns (GetResponse);

  rpc Put(PutRequest) returns (PutResponse);

  rpc Delete(DeleteRequest) returns (DeleteResponse);
}
```

In contrast to the messages defined in Example 8-2, the `rpc` definitions represent functions (verbs) which will send and receive messages (nouns).

In this example, we add three methods to our service:

- `Get`, which accepts a `GetRequest` and returns a `GetResponse`
- `Put`, which accepts a `PutRequest` and returns a `PutResponse`
- `Delete`, which accepts a `DeleteRequest` and returns a `DeleteResponse`

Note that we don't actually implement the functionality here. We do that later.

The previous methods are all examples of *unary RPC* definitions, in which a client sends a single request to the server and gets a single response back. This is the simplest of the four service methods types. Various streaming modes are also supported, but these are beyond the scope of this simple primer. The gRPC documentation discusses these in more detail.

Compiling your protocol buffers

Now that you have a *.proto* file complete with message and service definitions, the next thing you need to do is generate the classes you'll need to read and write messages. To do this, you need to run the protocol buffer compiler `protoc` on our *keyvalue.proto*.

If you haven't installed the `protoc` compiler and Go protocol buffers plug-in, follow the directions in "Installing the protocol compiler" on page 231 to do so.

Now you can run the compiler, specifying the source directory (`$SOURCE_DIR`) where your application's source code lives (which defaults to the the current directory), the destination directory (`$DEST_DIR`; often the same as `$SOURCE_DIR`), and the path to your *keystore.proto*. Because we want Go code, you use the `--go_out` option. `protoc` provides equivalent options to generate code for other supported languages as well.

In this case, we would invoke:

```
$ protoc --proto_path=$SOURCE_DIR \
    --go_out=$DEST_DIR --go_opt=paths=source_relative \
    --go-grpc_out=$DEST_DIR --go-grpc_opt=paths=source_relative \
    $SOURCE_DIR/keyvalue.proto
```

The `go_opt` and `go-grpc_opt` flags tell `protoc` to place the output files in the same relative directory as the input file. Our *keyvalue.proto* file results in two files, named *keyvalue.pb.go* and *keyvalue_grpc.pb.go*.

Without these flags, the output files are placed in a directory named after the Go package's import path. Our *keyvalue.proto* file, for example, would result in a file named `github.com/cloud-native-go/ch08/keyvalue/keyvalue.pb.go`.

Implementing the gRPC service

To implement our gRPC server, we'll need to implement the generated service interface, which defines the server API for our key-value service. It can be found in *keyvalue_grpc.pb.go* as `KeyValueServer`:

```
type KeyValueServer interface {
    Get(context.Context, *GetRequest) (*GetResponse, error)
    Put(context.Context, *PutRequest) (*PutResponse, error)
    Delete(context.Context, *DeleteRequest) (*PutResponse, error)
}
```

As you can see, the KeyValueServer interface specifies our Get, Put, and Delete methods: each accepts a context.Context and a request pointer, and returns a response pointer and an error.

> As a side effect of its simplicity, it's dead easy to mock requests to, and responses from, a gRPC server implementation.

To implement our server, we'll make use of a generated struct that provides a default implementation for the KeyValueServer interface, which, in our case, is named UnimplementedKeyValueServer. It's so named because it includes default "unimplemented" versions of all of our client methods attached, which look something like the following:

```
type UnimplementedKeyValueServer struct {}

func (*UnimplementedKeyValueServer) Get(context.Context, *GetRequest)
        (*GetResponse, error) {

    return nil, status.Errorf(codes.Unimplemented, "method not implemented")
}
```

By embedding the UnimplementedKeyValueServer, we're able to implement our key-value gRPC server. This is demonstrated with the following code, in which we implement the Get method. The Put and Delete methods are omitted for brevity:

```
package main

import (
    "context"
    "log"
    "net"

    pb "github.com/cloud-native-go/ch08/keyvalue"
    "google.golang.org/grpc"
)

// server is used to implement KeyValueServer. It MUST embed the generated
// struct pb.UnimplementedKeyValueServer
type server struct {
    pb.UnimplementedKeyValueServer
}

func (s *server) Get(ctx context.Context, r *pb.GetRequest)
        (*pb.GetResponse, error) {
```

```
    log.Printf("Received GET key=%v", r.Key)

    // The local Get function is implemented back in Chapter 5
    value, err := Get(r.Key)

    // Return expects a GetResponse pointer and an err
    return &pb.GetResponse{Value: value}, err
}

func main() {
    // Create a gRPC server and register our KeyValueServer with it
    s := grpc.NewServer()
    pb.RegisterKeyValueServer(s, &server{})

    // Open a listening port on 50051
    lis, err := net.Listen("tcp", ":50051")
    if err != nil {
        log.Fatalf("failed to listen: %v", err)
    }

    // Start accepting connections on the listening port
    if err := s.Serve(lis); err != nil {
        log.Fatalf("failed to serve: %v", err)
    }
}
```

In the previous code, we implement and start our service in four steps:

1. *Create the server struct.* Our `server` struct embeds `pb.UnimplementedKeyValueServer`. This is not optional: gRPC requires your server struct to similarly embed its generated `UnimplementedXXXServer`.
2. *Implement the service methods.* We implement the service methods defined in the generated `pb.KeyValueServer` interface. Interestingly, because the `pb.UnimplementedKeyValueServer` includes stubs for all of these service methods, we don't have to implement them all right away.
3. *Register our gRPC server.* In the `main` function, we create a new instance of the `server` struct and register it with the gRPC framework. This is similar to how we registered handler functions in "Building an HTTP Server with net/http" on page 112, except we register an entire instance rather than individual functions.
4. *Start accepting connections.* Finally, we open a listening port[7] using `net.Listen`, which we pass to the gRPC framework via `s.Serve` to begin listening.

7 If you wanted to be creative, this could be a `FileListener`, or even a `stdio` stream.

It could be argued that gRPC provides the best of both worlds by providing the freedom to implement any desired functionality without having to be concerned with building many of the tests and checks usually associated with a RESTful service.

Implementing the gRPC client

Because all of the client code is generated, making use of a gRPC client is fairly straightforward.

The generated client interface will be named `XXXClient`, which in our case will be `KeyValueClient`, shown in the following:

```go
type KeyValueClient interface {
    Get(ctx context.Context, in *GetRequest, opts ...grpc.CallOption)
        (*GetResponse, error)

    Put(ctx context.Context, in *PutRequest, opts ...grpc.CallOption)
        (*PutResponse, error)

    Delete(ctx context.Context, in *DeleteRequest, opts ...grpc.CallOption)
        (*PutResponse, error)
}
```

All of the methods described in our source `.proto` file are specified here, each accepting a request type pointer, and returning a response type pointer and an error.

Additionally, each of the methods accepts a `context.Context` (if you're rusty on what this is or how it's used, take a look at "The Context Package" on page 72), and zero or more instances of `grpc.CallOption`. `CallOption` is used to modify the behavior of the client when it executes its calls. More detail can be found in the gRPC API documentation.

I demonstrate how to create and use a gRPC client in the following:

```go
package main

import (
    "context"
    "log"
    "os"
    "strings"
    "time"

    pb "github.com/cloud-native-go/ch08/keyvalue"
    "google.golang.org/grpc"
)

func main() {
    // Set up a connection to the gRPC server
    conn, err := grpc.Dial("localhost:50051",
        grpc.WithInsecure(), grpc.WithBlock(), grpc.WithTimeout(time.Second))
    if err != nil {
        log.Fatalf("did not connect: %v", err)
    }
    defer conn.Close()

    // Get a new instance of our client
    client := pb.NewKeyValueClient(conn)

    var action, key, value string

    // Expect something like "set foo bar"
    if len(os.Args) > 2 {
        action, key = os.Args[1], os.Args[2]
        value = strings.Join(os.Args[3:], " ")
    }

    // Use context to establish a 1-second timeout.
    ctx, cancel := context.WithTimeout(context.Background(), time.Second)
    defer cancel()

    // Call client.Get() or client.Put() as appropriate.
    switch action {
    case "get":
        r, err := client.Get(ctx, &pb.GetRequest{Key: key})
        if err != nil {
            log.Fatalf("could not get value for key %s: %v\n", key, err)
        }
        log.Printf("Get %s returns: %s", key, r.Value)

    case "put":
        _, err := client.Put(ctx, &pb.PutRequest{Key: key, Value: value})
        if err != nil {
            log.Fatalf("could not put key %s: %v\n", key, err)
        }
        log.Printf("Put %s", key)

    default:
        log.Fatalf("Syntax: go run [get|put] KEY VALUE...")
    }
}
```

The preceding example parses command-line values to determine whether it should do a `Get` or a `Put` operation.

First, it establishes a connection with the gRPC server using the `grpc.Dial` function, which takes a target address string, and one or more `grpc.DialOption` arguments that configure how the connection gets set up. In our case we use:

- `WithInsecure`, which disables transport security for this `ClientConn`. *Don't use insecure connections in production.*
- `WithBlock`, which makes `Dial` a block until a connection is established, otherwise the connection will occur in the background.
- `WithTimeout`, which makes a blocking `Dial` throw an error if it takes longer than the specified amount of time.

Next, it uses `NewKeyValueClient` to get a new `KeyValueClient`, and gets the various command-line arguments.

Finally, based on the `action` value, we call either `client.Get` or `client.Put`, both of which return an appropriate return type and an error.

Once again, these functions look and feel exactly like local function calls. No checking status codes, hand-building our own clients, or any other funny business.

Loose Coupling Local Resources with Plug-ins

At first glance, the topic of loose coupling of local—as opposed to remote or distributed—resources might seem mostly irrelevant to a discussion of "cloud native" technologies. But you might be surprised how often such patterns come in handy.

For example, it's often useful to build services or tools that can accept data from different kinds of input sources (such as REST interface, a gRPC interface, and a chatbot interface) or generate different kinds of outputs (such as generating different kinds of logging or metric formats). As an added bonus, designs that support such modularity can also make mocking resources for testing dead simple.

As we'll see in "Hexagonal Architecture" on page 255, entire software architectures have even been built around this concept.

No discussion of loose coupling would be complete without a review of plug-in technologies.

In-Process Plug-ins with the plugin Package

Go provides a native plug-in system in the form of the standard `plugin` package. This package is used to open and access Go plug-ins, but it's not necessary to actually build the plug-ins themselves.

As we'll demonstrate in the following, the requirements for building and using a Go plug-in are pretty minimal. It doesn't have to even know it's a plug-in or even import the `plugin` package. A Go plug-in has three real requirements: it must be in the `main` package, it must export one or more functions or variables, and it must be compiled using the `-buildmode=plugin` build flag. That's it, really.

Go Plug-in Caveats

Before we get too deep into the subject of Go plug-ins, it's important to mention some caveats up front.

1. As of Go version 1.16.1, Go plug-ins are only supported on Linux, FreeBSD, and MacOS.
2. The version of Go used to build a plug-in must match the program that's using it *exactly*. Plug-ins built with Go 1.16.0 won't work with Go 1.16.1.
3. Similarly, the versions of any packages used by both the plug-in and program must also match *exactly*.
4. Finally, building plug-ins forces `CGO_ENABLED`, making cross-compiling more complicated.

These conditions make Go plug-ins most appropriate when you're building plug-ins for use within the same codebase, but it adds considerable obstacles for creating distributable plug-ins.

Plug-in vocabulary

Before we continue, we need to define a few terms that are particular to plug-ins. Each of the following describes a specific plug-in concept, and each has a corresponding type or function implementation in the `plugin` package. We'll go into all of these in more detail in our example.

Plug-in
: A *plug-in* is a Go `main` package with one or more exported functions and variables that has been built with the `-buildmode=plugin` build flag. It's represented in the `plugin` package by the `Plugin` type.

Open
: *Opening* a plug-in is the process of loading it into memory, validating it, and discovering its exposed symbols. A plug-in at a known location in the file system can be opened using the `Open` function, which returns a `*Plugin` value:

```go
func Open(path string) (*Plugin, error)
```

Symbol
: A plug-in *symbol* is any variable or function that's exported by the plug-in's package. Symbols can be retrieved by "looking them up," and are represented in the `plugin` package by the `Symbol` type:

```go
type Symbol interface{}
```

Look up

Looking up describes the process of searching for and retrieving a symbol exposed by a plug-in. The `plugin` package's `Lookup` method provides that functionality, and returns a `Symbol` value:

```
func (p *Plugin) Lookup(symName string) (Symbol, error)
```

In the next section, we present a toy example that demonstrates how these resources are used, and dig into a little detail in the process.

A toy plug-in example

You can only learn so much from a review of the API, even one as minimal as the `plugin` package. So, let's build ourselves a toy example: a program that tells you about various animals,[8] as implemented by plug-ins.

For this example, we'll be creating three independent packages with the following package structure:

```
~/cloud-native-go/ch08/go-plugin
├── duck
│   └── duck.go
├── frog
│   └── frog.go
└── main
    └── main.go
```

The `duck/duck.go` and `frog/frog.go` files each contain the source code for one plug-in. The `main/main.go` file contains our example's `main` function, which will load and use the plug-ins we'll generate by building `frog.go` and `duck.go`.

The complete source code for this example is available in this book's companion GitHub repository.

The Sayer interface

In order for a plug-in to be useful, the functions that access it need to know what symbols to look up and what contract those symbols conform to.

One convenient—but by no means required—way to do this is to use an interface that a symbol can be expected to satisfy. In our particular implementation our plug-ins will expose just one symbol—`Animal`—which we'll expect to conform to the following `Sayer` interface:

8 Yes, I know the animal thing has been done before. Sue me.

```
type Sayer interface {
    Says() string
}
```

This interface describes only one method, Says, which returns a string that says what an animal says.

The Go plugin code

We have source for two separate plug-ins in duck/duck.go and frog/frog.go. In the following snippet, the first of these, duck/duck.go, is shown in its entirety and displays all of the requirements of a plug-in implementation:

```
package main

type duck struct{}

func (d duck) Says() string {
    return "quack!"
}

// Animal is exported as a symbol.
var Animal duck
```

As described in the introduction to this section, the requirements for Go plug-in are really, really minimal: it just has to be a main package that exports one or more variables or functions.

The previous plug-in code describes and exports just one feature—Animal—that satisfies the preceding Sayer interface. Recall that exported package variables and symbols are exposed on the plug-in as shared library symbols that can be looked up later. In this case, our code will have to look specifically for the exported Animal symbol.

In this example we have only one symbol, but there's no explicit limit to the number of symbols we can have. We could have exported many more features, if we wanted to.

We won't show the frog/frog.go file here because it's essentially the same. But it's important to know that the internals of a plug-in don't matter as long as it satisfies the expectations of its consumer. These expectations are that:

- The plug-in exposes a symbol named Animal.
- The Animal symbol adheres to the contract defined by the Sayer interface.

Building the plug-ins

Building a Go plug-in is very similar to building any other Go main package, except that you have to include the -buildmode=plugin build parameter.

To build our `duck/duck.go` plug-in code, we do the following:

```
$ go build -buildmode=plugin -o duck/duck.so duck/duck.go
```

The result is a shared object (`.so`) file in ELF (Executable Linkable Format) format:

```
$ file duck/duck.so
duck/duck.so: Mach-O 64-bit dynamically linked shared library x86_64
```

ELF files are commonly used for plug-ins because once they're loaded into memory by the kernel they expose symbols in a way that allows for easy discovery and access.

Using our go plug-ins

Now that we've built our plug-ins, which are patiently sitting there with their `.so` extensions, we need to write some code that'll load and use them.

Note that even though we have our plug-ins fully built and in place, we haven't had to reach for the `plugin` package yet. However, now that we want to actually use our plug-ins, we get to change that now.

The process of finding, opening, and consuming a plug-in requires several steps, which I demonstrate next.

Import the plugin package. First things first: we have to import the `plugin` package, which will provide us the tools we need to open and access our plug-ins.

In this example, we import four packages: `fmt`, `log`, `os`, and most relevant to this example, `plugin`:

```go
import (
    "fmt"
    "log"
    "os"
    "plugin"
)
```

Find our plug-in. To load a plug-in, we have to find its relative or absolute file path. For this reason, plug-in binaries are usually named according to some pattern and placed somewhere where they can be easily discovered, like the user's command path or other standard fixed location.

For simplicity, our implementation assumes that our plug-in has the same name as the user's chosen animal and lives in a path relative to the execution location:

```go
if len(os.Args) != 2 {
    log.Fatal("usage: run main/main.go animal")
}

// Get the animal name, and build the path where we expect to
// find the corresponding shared object (.so) file.
```

```go
    name := os.Args[1]
    module := fmt.Sprintf("./%s/%s.so", name, name)
```

Importantly, this approach means that our plug-in doesn't need to be known—or even exist—at compile time. In this manner, we're able to implement whatever plug-ins we want at any time, and load and access them dynamically as we see fit.

Open our plug-in. Now that we think we know our plug-in's path we can use the `Open` function to "open" it, loading it into memory and discovering its available symbols. The `Open` function returns a `*Plugin` value that can then be used to look up any symbols exposed by the plug-in:

```go
    // Open our plugin and get a *plugin.Plugin.
    p, err := plugin.Open(module)
    if err != nil {
        log.Fatal(err)
    }
```

When a plug-in is first opened by the `Open` function, the `init` functions of all packages that aren't already part of the program are called. The package's `main` function is *not* run.

When a plug-in is opened, a single canonical `*Plugin` value representation of it is loaded into memory. If a particular path has already been opened, subsequent calls to `Open` will return the same `*Plugin` value.

A plug-in can't be loaded more than once, and can't be closed.

Look up your symbol. To retrieve a variable or function exported by our package—and therefore exposed as a symbol by the plug-in—we have to use the `Lookup` method to find it. Unfortunately, the `plugin` package doesn't provide any way to list all of the symbols exposed by a plug-in, so you we to know the name of our symbol ahead of time:

```go
    // Lookup searches for a symbol named "Animal" in plug-in p.
    symbol, err := p.Lookup("Animal")
    if err != nil {
        log.Fatal(err)
    }
```

If the symbol exists in the plug-in `p`, then `Lookup` returns a `Symbol` value. If the symbol doesn't exist in `p`, then a non-nil `error` is returned instead.

Assert and use your symbol. Now that we have our `Symbol`, we can convert it into the form we need and use it however we want. To make things nice and easy for us, the `Symbol` type is essentially a rebranded `interface{}` value. From the `plugin` source code:

```go
    type Symbol interface{}
```

This means that as long as we know what our symbol's type is, we can use type assertion to coerce it into a concrete type value that can be used however we see fit:

```
// Asserts that the symbol interface holds a Sayer.
animal, ok := symbol.(Sayer)
if !ok {
    log.Fatal("that's not a Sayer")
}

// Now we can use our loaded plug-in!
fmt.Printf("A %s says: %q\n", name, animal.Says())
```

In the previous code, we assert that the `symbol` value satisfies the `Sayer` interface. If it does, we print what our animal says. If it doesn't we're able to exit gracefully.

Executing our example

Now that we've written our main code that attempts to open and access the plug-in, we can run it like any other Go `main` package, passing the animal name in the arguments:

```
$ go run main/main.go duck
A duck says: "quack!"

$ go run main/main.go frog
A frog says: "ribbit!"
```

We can even implement arbitrary plug-ins later without changing our main source code:

```
$ go run main/main.go fox
A fox says: "ring-ding-ding-ding-dingeringeding!"
```

HashiCorp's Go Plug-in System over RPC

HashiCorp's Go plugin system has been in wide use—both internally to HashiCorp and elsewhere—since at least 2016, predating the release of Go's standard `plugin` package by about a year.

Unlike Go plug-ins, which use shared libraries, HashiCorp's plug-ins are standalone processes that are executed by using `exec.Command`, which has some obvious benefits over shared libraries:

They can't crash your host process
: Because they're separate processes, a `panic` in a plug-in doesn't automatically crash the plug-in consumer.

They're more version-flexible
: Go plug-ins are famously version-specific. HashiCorp plug-ins are far less so, expecting only that plug-ins adhere to a contract. It also supports explicit protocol versioning.

They're relatively secure
: HashiCorp plug-ins only have access to the interfaces and parameters passed to them, as opposed to the entire memory space of the consuming process.

They do have a couple of downsides, though:

More verbose
: HashiCorp plug-ins require more boilerplate than Go plug-ins.

Lower performance
: Because all data exchange with HashiCorp plug-ins occurs over RPC, communication with Go plug-ins is generally more performant.

That being said, let's take a look at what it takes to assemble a simple plug-in.

Another toy plug-in example

So we can compare apples to apples, we're going to work through a toy example that's functionally identical to the one for the standard `plugin` package in "A toy plug-in example" on page 243: a program that tells you what various animals say.

As before, we'll be creating several independent packages with the following structure:

```
~/cloud-native-go/ch08/hashicorp-plugin
├── commons
│   └── commons.go
├── duck
│   └── duck.go
└── main
    └── main.go
```

As before, the `duck/duck.go` file contains the source code for a plug-in, and the `main/main.go` file contains our example's `main` function that loads and uses the plug-in. Because both of these are independently compiled to produce executable binaries, both files are in the `main` package.

The `commons` package is new. It contains some resources that are shared by the plug-in and the consumer, including the service interface and some RPC boilerplate.

As before, the complete source code for this example is available in this book's companion GitHub repository.

Common code

The `commons` package contains some resources that are shared by both the plug-in and the consumer, so in our example it's imported by both the plug-in and client code.

It contains the RPC stubs that are used by the underlying `net/rpc` machinery to define the service abstraction for the host and allow the plug-ins to construct their service implementations.

The Sayer interface. The first of these is the `Sayer` interface. This is our service interface, which provides the service contract that the plug-in service implementations must conform to and that the host can expect.

It's identical to the interface that we used in "The Sayer interface" on page 243:

```
type Sayer interface {
    Says() string
}
```

The `Sayer` interface only describes one method: `Says`. Although this code is shared, as long as this interface doesn't change, the shared contract will be satisfied and the degree of coupling is kept fairly low.

The SayerPlugin struct. The more complex of the common resources is the `SayerPlugin` struct, shown in the following. It's an implementation of `plugin.Plugin`, the primary plug-in interface from the `github.com/hashicorp/go-plugin` package.

The package declaration inside the *github.com/hashicorp/go-plugin* repository is `plugin`, not `go-plugin`, as its path might suggest. Adjust your imports accordingly!

The `Client` and `Server` methods are used to describe our service according to the expectations of Go's standard `net/rpc` package. We won't cover that package in this book, but if you're interested, you can find a wealth of information in the Go documentation:

```
type SayerPlugin struct {
    Impl Sayer
}

func (SayerPlugin) Client(b *plugin.MuxBroker, c *rpc.Client)
        (interface{}, error) {

    return &SayerRPC{client: c}, nil
}
```

```
func (p *SayerPlugin) Server(*plugin.MuxBroker) (interface{}, error) {
    return &SayerRPCServer{Impl: p.Impl}, nil
}
```

Both methods accept a `plugin.MuxBroker`, which is used to create multiplexed streams on a plug-in connection. While very useful, this is a more advanced use case that we won't have time to cover in this book.

The SayerRPC client implementation. SayerPlugin's `Client` method provides an implementation of our `Sayer` interface that communicates over an RPC client—the appropriately named `SayerRPC` struct—shown in the following:

```
type SayerRPC struct{ client *rpc.Client }

func (g *SayerRPC) Says() string {
    var resp string

    err := g.client.Call("Plugin.Says", new(interface{}), &resp)
    if err != nil {
        panic(err)
    }

    return resp
}
```

`SayerRPC` uses Go's RPC framework to remotely call the `Says` method implemented in the plug-in. It invokes the `Call` method attached to the `*rpc.Client`, passing in any parameters (`Says` doesn't have any parameters, so we pass an empty `interface{}`) and retrieves the response, which it puts it into the `resp` string.

The handshake configuration. `HandshakeConfig` is used by both the plug-in and host to do a basic handshake between the host and the plug-in. If the handshake fails—if the plug-in was compiled with a different protocol version, for example—a user-friendly error is shown. This prevents users from executing bad plug-ins or executing a plug-in directly. Importantly, this is a UX feature, not a security feature:

```
var HandshakeConfig = plugin.HandshakeConfig{
    ProtocolVersion:  1,
    MagicCookieKey:   "BASIC_PLUGIN",
    MagicCookieValue: "hello",
}
```

The SayerRPCServer server implementation. SayerPlugin's `Server` method provides a definition of an RPC server—the `SayerRPCServer` struct—to serve the actual methods in a way that's consistent with `net/rpc`:

```
type SayerRPCServer struct {
    Impl Sayer    // Impl contains our actual implementation
}

func (s *SayerRPCServer) Says(args interface{}, resp *string) error {
    *resp = s.Impl.Says()
    return nil
}
```

SayerRPCServer doesn't implement the Sayer service. Instead, its Says method calls into a Sayer implementation—Impl—that we'll provide when we use this to build our plug-in.

Our plug-in implementation

Now that we've assembled the code that's common between the host and plug-ins—the Sayer interface and the RPC stubs—we can build our plug-in code. The code in this section represents the entirety of our main/main.go file.

Just like standard Go plug-ins, HashiCorp plug-ins are compiled into standalone executable binaries, so they must be in the main package. Effectively, every HashiCorp plug-in is a small, self-contained RPC server:

```
package main
```

We have to import our commons package, as well as the hashicorp/go-plugin package, whose contents we'll reference as plugin:

```
import (
    "github.com/cloud-native-go/ch08/hashicorp-plugin/commons"
    "github.com/hashicorp/go-plugin"
)
```

In our plug-ins we get to build our real implementations. We can build it however we want,[9] as long as it conforms to the Sayer interface that we define in the commons package:

```
type Duck struct{}

func (g *Duck) Says() string {
    return "Quack!"
}
```

Finally, we get to our main function. It's somewhat "boilerplate-y" but it's essential:

```
func main() {
    // Create and initialize our service implementation.
    sayer := &Duck{}

    // pluginMap is the map of plug-ins we can dispense.
```

9 So, naturally, we're building a duck. Obviously.

```
    var pluginMap = map[string]plugin.Plugin{
        "sayer": &commons.SayerPlugin{Impl: sayer},
    }

    plugin.Serve(&plugin.ServeConfig{
        HandshakeConfig: handshakeConfig,
        Plugins:         pluginMap,
    })
}
```

The `main` function does three things. First, it creates and initializes our service implementation, a `*Duck` value, in this case.

Next, it maps the service implementation to the name "sayer" in the `pluginMap`. If we wanted to, we could actually implement several plug-ins, listing them all here with different names.

Finally, we call `plugin.Serve`, which starts the RPC server that will handle any connections from the host process, allowing the handshake with the host to proceed and the service's methods to be executed as the host sees fit.

Our host process

We now have our host process; the main command that acts as a client that finds, loads, and executes the plug-in processes.

As you'll see, using HashiCorp plug-ins isn't all that different from the steps that described for Go plug-ins in "Using our go plug-ins" on page 245.

Import the hashicorp/go-plugin and commons packages. As usual, we start with our package declaration and imports. The imports mostly aren't interesting, and their necessity should be clear from examination of the code.

The two that *are* interesting (but not surprising) are `github.com/hashicorp/go-plugin`, which we, once again, have to reference as `plugin`, and our `commons` package, which contains the interface and handshake configuration, both of which must be agreed upon by the host and the plug-ins:

```
package main

import (
    "fmt"
    "log"
    "os"
    "os/exec"

    "github.com/cloud-native-go/ch08/hashicorp-plugin/commons"
    "github.com/hashicorp/go-plugin"
)
```

Find our plug-in. Since our plug-in is an external file, we have to find it. Again, for simplicity, our implementation assumes that our plug-in has the same name as the user's chosen animal and lives in a path relative to the execution location:

```
func main() {
    if len(os.Args) != 2 {
        log.Fatal("usage: run main/main.go animal")
    }

    // Get the animal name, and build the path where we expect to
    // find the corresponding executable file.
    name := os.Args[1]
    module := fmt.Sprintf("./%s/%s", name, name)

    // Does the file exist?
    _, err := os.Stat(module)
    if os.IsNotExist(err) {
        log.Fatal("can't find an animal named", name)
    }
}
```

It bears repeating that the value of this approach is that our plug-in—and its implementation—doesn't need to be known, or even exist, at compile time. We're able to implement whatever plug-ins we want at any time, and use then dynamically as we see fit.

Create our plug-in client. The first way that a HashiCorp RPC plug-in differs from a Go plug-in is the way that it retrieves the implementation. Where Go plug-ins have to be "opened" and their symbol "looked up," HashiCorp plug-ins are built on RPC, and therefore require an RPC client.

This actually requires two steps, and two clients: a `*plugin.Client` that manages the lifecycle of the plug-in subprocess, and a protocol client—a `plugin.ClientProtocol` implementation—that can communicate with the plug-in subprocess.

This awkward API is mostly historical, but is used to split the client that deals with subprocess management and the client that does RPC management:

```
// pluginMap is the map of plug-ins we can dispense.
var pluginMap = map[string]plugin.Plugin{
    "sayer": &commons.SayerPlugin{},
}

// Launch the plugin process!
client := plugin.NewClient(&plugin.ClientConfig{
    HandshakeConfig: commons.HandshakeConfig,
    Plugins:         pluginMap,
    Cmd:             exec.Command(module),
})
defer client.Kill()
```

```
    // Connect to the plugin via RPC
    rpcClient, err := client.Client()
    if err != nil {
        log.Fatal(err)
    }
```

Most of this snippet consists of defining the parameters of the plug-in that we want in the form of a `plugin.ClientConfig`. The complete list of available client configurations is lengthy. This example uses only three:

`HandshakeConfig`
: The handshake configuration. This has to match the plug-in's own handshake configuration or we'll get an error in the next step.

`Plugins`
: A map that specifies the name and type of the plug-in we want.

`Cmd`
: An `*exec.Cmd` value that represents the command for starting the plug-in subprocess.

With all of the configuration stuff out of the way, we can first use `plugin.NewClient` to retrieve a `*plugin.Client` value, which we call `client`.

Once we have that, we can use `client.Client` to request a protocol client. We call this `rpcClient` because it knows how to use RPC to communicate with the plug-in subprocess.

Connect to our plug-in and dispense our Sayer. Now that we have our protocol client, we can use it to dispense our `Sayer` implementation:

```
        // Request the plug-in from the client
        raw, err := rpcClient.Dispense("sayer")
        if err != nil {
            log.Fatal(err)
        }

        // We should have a Sayer now! This feels like a normal interface
        // implementation, but is actually over an RPC connection.
        sayer := raw.(commons.Sayer)

        // Now we can use our loaded plug-in!
        fmt.Printf("A %s says: %q\n", name, sayer.Says())
    }
```

Using the protocol client's `Dispense` function, we're able to finally retrieve our `Sayer` implementation as an `interface{}`, which we can assert as a `commons.Sayer` value and immediately use exactly like using a local value.

Under the covers, our `sayer` is in fact a `SayerRPC` value, and calls to its functions trigger RPC calls that are executed in our plug-in's address space.

In the next section, we'll introduce the hexagonal architecture, an architectural pattern built around the entire concept of loose coupling by using easily exchangeable "ports and adapters" to connect to its environment.

Hexagonal Architecture

Hexagonal architecture—also known as the "ports and adapters" pattern—is an architectural pattern that uses loose coupling and *inversion of control* as its central design philosophy to establish clear boundaries between business and peripheral logic.

In a hexagonal application, the core application doesn't know any details at all about the outside world, operating entirely through loosely coupled *ports* and technology-specific *adapters*.

This approach allows the application to, for example, expose different APIs (REST, gRPC, a test harness, etc.) or use different data sources (database, message queues, local files, etc.) without impacting its core logic or requiring major code changes.

It took me an embarrassingly long time to realize that the name "hexagonal architecture" doesn't actually mean anything. Alistair Cockburn, the author of hexagonal architecture, chose the shape because it gave him enough room to illustrate the design.

The Architecture

As illustrated in Figure 8-3, hexagonal architecture is composed of three components conceptually arranged in and around a central hexagon:

The core application
: The application proper, represented by the hexagon. This contains all of the business logic but has no direct reference to any technology, framework, or real world device. The business logic shouldn't depend on whether it exposes a REST or a gRPC API, or whether it gets data from a database or a *.csv* file. Its only view of the world should be through ports.

Ports and adapters
: The ports and adapters are represented on the edge of the hexagon. Ports allow different kinds of actors to "plug in" and interact with the core service. Adapters can "plug into" a port and translate signals between the core application and an actor.

For example, your application might have a "data port" into which a "data adapter" might plug. One data adapter might write to a database, while another might use an in-memory datastore or automated test harness.

Actors
The actors can be anything in the environment that interacts with the core application (users, upstream services, etc.) or that the core application interacts with (storage devices, downstream services, etc.). They exist outside the hexagon.

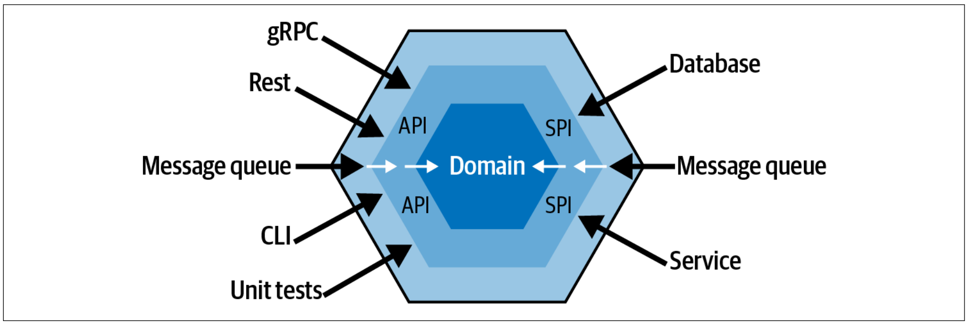


Figure 8-3. All dependencies in the hexagonal architecture point inward; the hexagons represent the core application's domain and API layers, and the ports and adapters are represented as arrows on the edge of the hexagon, each of which interfaces with a particular actor

In a traditional layered architecture, all of the dependencies point in the same direction, with each layer depending on the one below it.

In a hexagonal architecture, however, all dependencies point inward: the core business logic doesn't know any details about the outer world, the adapters know how to ferry information to and from the core, and the adapters in the outer world know how to interact with the actors.

Implementing a Hexagonal Service

To illustrate this, we're going to refactor our old friend the key-value store.

If you'll recall from Chapter 5, the core application of our key-value store reads and writes to a in-memory map, which can be accessed via a RESTful (or gRPC) frontend. Later in the same chapter, we implemented a transaction logger, which knows how to write all transactions to *somewhere* and read them all back when the system restarts.

We'll reproduce important snippets of the service here, but if you want a refresher on what we did, go back and do so now.

By this point in the book, we've accumulated a couple of different implementations for a couple of different components of our service that seem like good candidates for ports and adapters in a hexagonal architecture:

The frontend
: Back in "Generation 1: The Monolith" on page 112 we implemented a REST frontend, and then in "Remote Procedure Calls with gRPC" on page 230 we implemented a separate gRPC frontend. We can describe these with a single "driver" port into which we'll be able to plug either (or both!) as adapters.

The transaction logger
: In "What's a Transaction Log?" on page 125 we created two implementations of a transaction log. These seem like a natural choice for a "driven" port and adapters.

While all the logic for all of these already exists, we'll need to do some refactoring to make this architecture "hexagonal":

1. Our original core application—originally described in "Generation 0: The Core Functionality" on page 110—uses exclusively public functions. We'll refactor those into struct methods to make it easier to use in a "ports and adapters" format.
2. Both the RESTful and gRPC frontends are already consistent with hexagonal architecture, since the core application doesn't know or care about them, but they're constructed in a `main` function. We'll convert these into `FrontEnd` adapters into which we can pass our core application. This pattern is typical of a "driver" port.
3. The transaction loggers themselves won't need much refactoring, but they're currently embedded in the frontend logic. When we refactor the core application, we'll add a transaction logger port so that the adapter can be passed into the core logic. This pattern is typical of a "driven" port.

In the next section, we'll begin taking the existing components and refactoring them in accordance with hexagonal principles.

Our refactored components

For the sake of this example, all of our components live under the `github.com/cloud-native-go/examples/ch08/hexarch` package:

```
~/cloud-native-go/ch08/hexarch/
├── core
│   └── core.go
├── frontend
│   ├── grpc.go
│   └── rest.go
```

```
    ├── main.go
    └── transact
        ├── filelogger.go
        └── pglogger.go
```

`core`
: The core key-value application logic. Importantly, it has no dependencies outside of the Go standard libraries.

`frontend`
: Contains the REST and gRPC frontend driver adapters. These have a dependency on `core`.

`transact`
: Contains the file and PostgreSQL transaction logger driven adapters. These also have a dependency on `core`.

`main.go`
: Makes the core application instance, into which it passes the driven components, and which it passes to the driver adapters.

The complete source code is also available in the companion GitHub repository.

Now that we have our very high-level structure, let's go ahead and implement our first plug.

Our first plug

You may remember that we also implemented a *transaction log* to maintain a record of every time a resource is modified so that if our service crashes, is restarted, or otherwise finds itself in an inconsistent state, it can reconstruct its complete state by replaying the transactions.

In "Your transaction logger interface" on page 126, we represented a generic transaction logger with the `TransactionLogger`:

```go
type TransactionLogger interface {
    WriteDelete(key string)
    WritePut(key, value string)
}
```

For brevity, we only define the `WriteDelete` and `WritePut` methods.

A common aspect of "driven" adapters is that the core logic acts *on them*, so the core application has to know about the port. As such, this code lives in the `core` package.

Our core application

In our original implementation in "Your Super Simple API" on page 111, the transaction logger was used by the frontend. In a hexagonal architecture we move the port—in the form of the `TransactionLogger` interface—into the core application:

```
package core

import (
    "errors"
    "log"
    "sync"
)

type KeyValueStore struct {
    m        map[string]string
    transact TransactionLogger
}

func NewKeyValueStore(tl TransactionLogger) *KeyValueStore {
    return &KeyValueStore{
        m:        make(map[string]string),
        transact: tl,
    }
}

func (store *KeyValueStore) Delete(key string) error {
    delete(store.m, key)
    store.transact.WriteDelete(key)
    return nil
}

func (store *KeyValueStore) Put(key string, value string) error {
    store.m[key] = value
    store.transact.WritePut(key, value)
    return nil
}
```

Comparing the previous code with the original form in "Generation 0: The Core Functionality" on page 110, you'll see some significant changes.

First, `Put` and `Delete` aren't pure functions anymore: they're now methods on a new `KeyValueStore` struct, which also has the map data structure. We've also added a `NewKeyValueStore` function that initializes and returns a new `KeyValueStore` pointer value.

Finally, `KeyValueStore` now has a `TransactionLogger`, which `Put` and `Delete` act upon appropriately. This is our port.

Our TransactionLogger adapters

In Chapter 5 we created two `TransactionLogger` implementations:

- In "Implementing your FileTransactionLogger" on page 129, we describe a file-based implementation.
- In "Implementing your PostgresTransactionLogger" on page 139, we describe a PostgreSQL-backed implementation.

Both of these have been moved to the `transact` package. They hardly have to change at all, except to account for the fact that the `TransactionLogger` interface and `Event` struct now live in the `core` package.

But how do we determine which one to load? Well, Go doesn't have annotations or any fancy dependency injection features,[10] but there are still a couple of ways you can do this.

The first option is to use plug-ins of some kind (this is actually a primary use case for Go plug-ins). This might make sense if you want changing adapters to require *zero* code changes.

More commonly, you'll see some kind of "factory" function[11] that's used by the initializing function. While this still requires code changes to add adapters, they're isolated to a single, easily modified location. A more sophisticated approach might accept a parameter or configuration value to choose which adapter to use.

An example of a `TransactionLogger` factory function might look like the following:

```
func NewTransactionLogger(logger string) (core.TransactionLogger, error) {
    switch logger {
    case "file":
        return NewFileTransactionLogger(os.Getenv("TLOG_FILENAME"))

    case "postgres":
        return NewPostgresTransactionLogger(
            PostgresDbParams{
                dbName: os.Getenv("TLOG_DB_HOST"),
                host: os.Getenv("TLOG_DB_DATABASE"),
                user: os.Getenv("TLOG_DB_USERNAME"),
                password: os.Getenv("TLOG_DB_PASSWORD"),
            }
        )

    case "":
```

10 Good riddance.

11 I'm sorry.

```
            return nil, fmt.Errorf("transaction logger type not defined")

        default:
            return nil, fmt.Errorf("no such transaction logger %s", s)
        }
    }
```

In this example, the `NewTransactionLogger` function accepts a string that specifies the desired implementation, returning either one of our implementations or an `error`. We use the `os.Getenv` function to retrieve the appropriate parameters from environment variables.

Our FrontEnd port

But what about our frontends? If you will recall, we now have two frontend implementations:

- In "Generation 1: The Monolith" on page 112 in Chapter 5, we built a RESTful interface using `net/http` and `gorilla/mux`.
- In "Remote Procedure Calls with gRPC" on page 230, earlier in this chapter, we built an RPC interface with gRPC.

Both of these implementations include a `main` function where we configure and start the service to listen for connections.

Since they're "driver" ports, we need to pass the core application to them, so let's refactor both frontends into structs according to the following interface:

```
package frontend

type FrontEnd interface {
    Start(kv *core.KeyValueStore) error
}
```

The `FrontEnd` interface serves as our "frontend port," which all frontend implementations are expected to satisfy. The `Start` method accepts the core application API in the form of a `*core.KeyValueStore`, and will also include the setup logic that formerly lived in a `main` function.

Now that we have this, we can refactor both frontends so that they comply with the `FrontEnd` interface, starting with the RESTful frontend. As usual, the complete source code for this and the gRPC service refactor are available in this book's companion GitHub repository:

```go
package frontend

import (
    "net/http"

    "github.com/cloud-native-go/examples/ch08/hexarch/core"
    "github.com/gorilla/mux"
)

// restFrontEnd contains a reference to the core application logic,
// and complies with the contract defined by the FrontEnd interface.
type restFrontEnd struct {
    store *core.KeyValueStore
}

// keyValueDeleteHandler handles the logic for the DELETE HTTP method.
func (f *restFrontEnd) keyValueDeleteHandler(w http.ResponseWriter,
        r *http.Request) {

    vars := mux.Vars(r)
    key := vars["key"]

    err := f.store.Delete(key)
    if err != nil {
        http.Error(w, err.Error(), http.StatusInternalServerError)
        return
    }
}

// ...other handler functions omitted for brevity.

// Start includes the setup and start logic that previously
// lived in a main function.
func (f *restFrontEnd) Start(store *core.KeyValueStore) error {
    // Remember our core application reference.
    f.store = store

    r := mux.NewRouter()

    r.HandleFunc("/v1/{key}", f.keyValueGetHandler).Methods("GET")
    r.HandleFunc("/v1/{key}", f.keyValuePutHandler).Methods("PUT")
    r.HandleFunc("/v1/{key}", f.keyValueDeleteHandler).Methods("DELETE")

    return http.ListenAndServe(":8080", r)
}
```

Comparing the previous code to the code we produced in "Generation 1: The Monolith" on page 112, some differences stand out:

- All functions are now methods attached to a `restFrontEnd` struct.
- All calls to the core application go through the `store` value that lives in the `restFrontEnd` struct.
- Creating the router, defining the handlers, and starting the server now live in the `Start` method.

Similar changes will have been made for our gRPC frontend implementation to make it consistent with the `FrontEnd` port.

This new arrangement makes it easier for a consumer to choose and plug in a "frontend adapter," as demonstrated in the following.

Putting it all together

Here, we have our `main` function, in which we plug all of the components into our application:

```go
package main

import (
    "log"

    "github.com/cloud-native-go/examples/ch08/hexarch/core"
    "github.com/cloud-native-go/examples/ch08/hexarch/frontend"
    "github.com/cloud-native-go/examples/ch08/hexarch/transact"
)

func main() {
    // Create our TransactionLogger. This is an adapter that will plug
    // into the core application's TransactionLogger port.
    tl, err := transact.NewTransactionLogger(os.Getenv("TLOG_TYPE"))
    if err != nil {
        log.Fatal(err)
    }

    // Create Core and tell it which TransactionLogger to use.
    // This is an example of a "driven agent"
    store := core.NewKeyValueStore(tl)
    store.Restore()

    // Create the frontend.
    // This is an example of a "driving agent."
    fe, err := frontend.NewFrontEnd(os.Getenv("FRONTEND_TYPE"))
    if err != nil {
        log.Fatal(err)
    }

    log.Fatal(fe.Start(store))
}
```

First, we then create a transaction logger according to the environment `TLOG_TYPE`. We do this first because the “transaction logger port” is “driven,” so we’ll need to provide it to the application to plug it in.

We then create our `KeyValueStore` value, which represents our core application functions and provides an API for ports to interact with, and provide it with any driven adapters.

Next, we create any “driver” adapters. Since these act on the core application API, we provide the API to the adapter instead of the other way around as we would with a “driven” adapter. This means we could also create multiple frontends here, if we wanted, by creating a new adapter and passing it the `KeyValueStore` that exposes the core application API.

Finally, we call `Start` on our frontend, which instructs it to start listening for connections. At last, we have a complete hexagonal service!

Summary

We covered a lot of ground in this chapter, but really only scratched the surface of all the different ways that components can find themselves tightly coupled and all the different ways of managing each of those tightly coupled components.

In the first half of the chapter, we focused on the coupling that can result from how services communicate. We talked about the problems caused by fragile exchange protocols like SOAP, and demonstrated REST and gRPC, which are less fragile because they can be changed to some degree without necessarily forcing client upgrades. We also touched on coupling “in time,” in which one service implicitly expects a timely response from another, and how publish-subscribe messaging might be used to relieve this.

In the second half we addressed some of the ways that systems can minimize coupling to local resources. After all, even distributed services are just programs, subject to the same limitations of the architectures and implementations as any program. Plug-ins implementations and hexagonal architectures are two ways of doing this by enforcing separation of concerns and inversion of control.

Unfortunately, we didn’t get to drill down into some other fascinating topics like service discovery, but, sadly, I had to draw a line somewhere before this subject got away from me!

CHAPTER 9

Resilience

A distributed system is one in which the failure of a computer you didn't even know about can render your own computer unusable.[1]

—Leslie Lamport, *DEC SRC Bulletin Board (May 1987)*

Late one September night, at just after two in the morning, a portion of Amazon's internal network quietly stopped working.[2] This event was brief, and not particularly interesting, except that it happened to affect a sizable number of the servers that supported the DynamoDB service.

Most days, this wouldn't be such a big deal. Any affected servers would just try to reconnect to the cluster by retrieving their membership data from a dedicated metadata service. If that failed, they would temporarily take themselves offline and try again.

But this time, when the network was restored, a small army of storage servers simultaneously requested their membership data from the metadata service, overwhelming it so that requests—even ones from previously unaffected servers—started to time out. Storage servers dutifully responded to the timeouts by taking themselves offline and retrying (again), further stressing the metadata service, causing even more servers to go offline, and so on. Within minutes, the outage had spread to the entire cluster. The service was effectively down, taking a number of dependent services down with it.

1 Lamport, Leslie. *DEC SRC Bulletin Board*, 28 May 1987. *https://oreil.ly/nD85V*.

2 Summary of the Amazon DynamoDB Service Disruption and Related Impacts in the US-East Region. Amazon AWS, September 2015. *https://oreil.ly/Y1P5S*.

To make matters worse, the sheer volume of retry attempts—a "retry storm"—put such a burden on the metadata service that it even became entirely unresponsive to requests to add capacity. The on-call engineers were forced to explicitly block requests to the metadata service just to relieve enough pressure to allow them to manually scale up.

Finally, nearly five hours after the initial network hiccup that triggered the incident, normal operations resumed, putting an end to what must have been a long night for all involved.

Keeping on Ticking: Why Resilience Matters

So, what was the root cause of Amazon's outage? Was it the network disruption? Was it the storage servers' enthusiastic retry behavior? Was it the metadata service's response time, or maybe its limited capacity?

Clearly, what happened that early morning didn't have a single root cause. Failures in complex systems never do.[3] Rather, the system failed as complex systems do: with a failure in a subsystem, which triggered a latent fault in another subsystem causing *it* to fail, followed by another, and another, until eventually the entire system went down. What's interesting, though, is that if any of the components in our story—the network, the storage servers, the metadata service—had been able to isolate and recover from failures elsewhere in the system, the overall system likely would have recovered without human intervention.

Unfortunately, this is just one example of a common pattern. Complex systems fail in complex (and often surprising) ways, but they don't fail all at once: they fail one subsystem at a time. For this reason, resilience patterns in complex systems take the form of bulwarks and safety valves that work to isolate failures at component boundaries. Frequently, a failure contained is a failure avoided.

This property, the measure of a system's ability to withstand and recover from errors and failures, is its *resilience*. A system can be considered *resilient* if it can continue operating correctly—possibly at a reduced level—rather than failing completely when one of its subsystems fails.

3 Cook, Richard I. "How Complex Systems Fail." 1998. *https://oreil.ly/WyJ4Q*.

Resilience Is Not Reliability

The terms *resilience* and *reliability* describe closely related concepts, and are often confused. But they aren't quite the same thing.[4]

- The *resilience* of a system is the degree to which it can continue to operate correctly in the face of errors and faults. Resilience, along with the other four cloud native properties, is just one factor that contributes to reliability.
- The *reliability* of a system is its ability to behave as expected for a given time interval. Reliability, in conjunction with attributes like availability and maintainability, contributes to a system's overall dependability.

What Does It Mean for a System to Fail?

> For want of a nail the shoe was lost,
> for want of a shoe the horse was lost;
> for want of a horse the rider was lost;
> all for want of care about a horse-shoe nail.
>
> —Benjamin Franklin, *The Way to Wealth (1758)*

If we want to know what it means for a system to fail, we first have to ask what a "system" is.

This is important. Bear with me.

By definition, a *system* is a set of components that work together to accomplish an overall goal. So far, so good. But here's the important part: each component of a system—a *subsystem*—is also a complete system unto itself, that in turn is composed of still smaller subsystems, and so on, and so on.

Take a car, for example. Its engine is one of dozens of subsystems, but it—like all the others—is also a very complex system with a number of subsystems of its own, including a cooling subsystem, which includes a thermostat, which includes a temperature switch, and so on. That's just some of thousands of components and subcomponents and sub-subcomponents. It's enough to make the mind spin: so many things that can fail. But what happens when they do?

As we mentioned earlier—and discussed in some depth in Chapter 6—failures of complex systems don't just happen all at once. They unravel in predictable steps:

4 If you're interested in a complete academic treatment, I highly recommend *Reliability and Availability Engineering* by Kishor S. Trivedi and Andrea Bobbio (Cambridge University Press).

1. All systems contain *faults*, which we lovingly refer to as “bugs” in the software world. A tendency for a temperature switch in a car engine to stick would be a fault. So would the metadata service’s limited capacity and the storage server’s retry behavior in the DynamoDB case study.[5] Under the right conditions, a fault can be exercised to produce an *error*.
2. An *error* is any discrepancy between the system’s intended and actual behavior. Many errors can be caught and handled appropriately, but if they’re not they can —singly or in accumulation—give rise to a *failure*. A stuck temperature switch in a car engine’s thermostat is an error.
3. Finally, a system can be said to be experiencing a *failure* when it’s no longer able to provide correct service.[6] A temperature switch that no longer responds to high temperatures can be said to have failed. A failure at the subsystem level becomes a fault at the system level.

This last bit bears repeating: *a failure at the subsystem level becomes a fault at the system level.* A stuck temperature switch causes a thermostat to fail, preventing coolant from flowing through the radiator, raising the temperature of the engine, causing it to stall and the car to stop.[7]

That’s how systems fail. It starts with the failure of one component—one subsystem —which causes an error in one or more components that interact with it, and ones that interact with that, and so on, propagating upward until the entire system fails.

This isn’t just academic. Knowing how complex systems fail—one component at a time—makes the means of resisting failures clearer: if a fault can be contained before it propagates all the way to the system level, the system may be able to recover (or at least fail on its own terms).

Building for Resilience

In a perfect world it would be possible to rid a system of every possible fault, but this isn’t realistic, and it’s wasteful and unproductive to try. By instead assuming that all components are destined to fail eventually—which they absolutely are—and designing them to respond gracefully to errors when they do occur, you can produce a system that’s functionally healthy even when some of its components are not.

There are lots of ways to increase the resiliency of a system. Redundancy, such as deploying multiple components of the same type, is probably the most common

5 Importantly, many faults are only evident in retrospect.

6 See? We eventually got there.

7 Go on, ask me how I know this.

approach. Specialized logic like circuit breakers and request throttles can be used to isolate specific kinds of errors, preventing them from propagating. Faulty components can even be reaped—or intentionally allowed to fail—to benefit the health of the larger system.

Resilience is a particularly rich subject. We'll explore several of these approaches—and more—over the remainder of the chapter.

Cascading Failures

The reason the DynamoDB case study is so appropriate is that it demonstrates so many different ways that things that can go wrong at scale.

Take, for example, how the failure of a group of storage servers caused requests to the metadata service to time out, which in turn caused more storage servers to fail, which increased the pressure on the metadata service, and so on. This is an excellent example of a particular—and particularly common—failure mode known as a *cascading failure*. Once a cascading failure has begun, it tends to spread very quickly, often on the order of a few minutes.

The mechanisms of cascading failures can vary a bit, but one thing they share is some kind of positive feedback mechanism. One part of a system experiences a local failure —a reduction in capacity, an increase in latency, etc.—that causes other components to attempt to compensate for the failed component in a way that exacerbates the problem, eventually leading to the failure of the entire system.

The classic cause of cascading failures is overload, illustrated in Figure 9-1. This occurs when one or more nodes in a set fails, causing the load to be catastrophically redistributed to the survivors. The increase in load overloads the remaining nodes, causing them to fail from resource exhaustion, taking the entire system down.

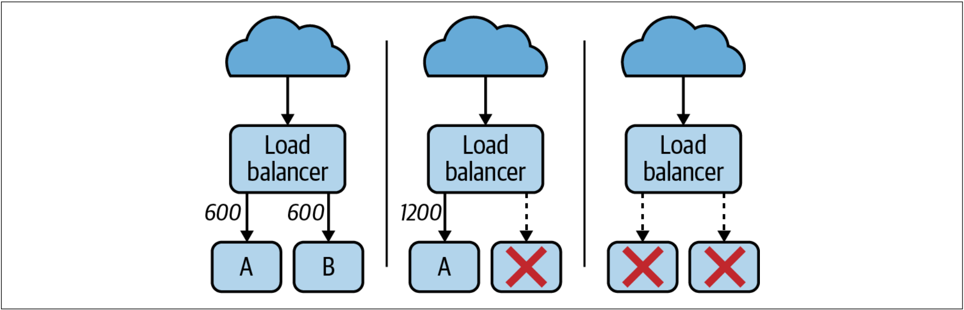


Figure 9-1. Server overload is a common cause of cascade failures; each server handles 600 requests per second, so when server B fails, server A is overloaded and also fails

The nature of positive feedback often makes it very difficult to scale your way out of a cascading failure by adding more capacity. New nodes can be overwhelmed as quickly as they come online, often contributing the feedback that took the system down in the first place. Sometimes, the only fix is to take your entire service down—perhaps by explicitly blocking the problematic traffic—in order to recover, and then slowly reintroduce load.

But how do you prevent cascading failures in the first place? This will be the subject of the next section (and, to some extent, most of this chapter).

Preventing Overload

Every service, however well-designed and implemented, has its functional limitations. This is particularly evident in services intended to handle and respond to client requests.[8] For any such service there exists some request frequency, a threshold beyond which bad things will start to happen. So, how do we keep a large number of requests from accidentally (or intentionally!) bringing our service down?

Ultimately, a service that finds itself in such a situation has no choice but to reject—partially or entirely—some number of requests. There are two main strategies for doing this:

Throttling
: Throttling is a relatively straightforward strategy that kicks in when requests come in faster than some predetermined frequency, typically, by just refusing to handle them. This is often used as a preventative measure by ensuring that no particular user consumes more resources than they would reasonably require.

Load shedding
: Load shedding is a little more adaptive. Services using this strategy intentionally drop ("shed") some proportion of load as they approach overload conditions by either refusing requests or falling back into a degraded mode.

These strategies aren't mutually exclusive; a service may choose to employ either or both of them, according to its needs.

Throttling

As we discussed in Chapter 4, a throttle pattern works a lot like the throttle in a car, except that instead of limiting the amount of fuel entering an engine, it limits the number of requests that a user (human or otherwise) can make to a service in a set period of time.

8 Especially if the service is available on the open sewer that is the public internet.

The general-purpose throttle example that we provided in "Throttle" on page 86 was relatively simple, and effectively global, at least as written. However, throttles are also frequently applied on a per-user basis to provide something like a usage quota, so that no one caller can consume too much of a service's resources.

In the following, we demonstrate a throttle implementation that, while still using a token bucket,[9] is otherwise quite different in several ways.

First, instead of having a single bucket that's used to gate all incoming requests, the following implementation throttles on a per-user basis, returning a function that accepts a "key" parameter, that's meant to represent a username or some other unique identifier.

Second, rather than attempting to "replay" a cached value when imposing a throttle limit, the returned function returns a Boolean that indicates when a throttle has been imposed. Note that the throttle doesn't return an `error` when it's activated: throttling isn't an error condition, so we don't treat it as one.

Finally, and perhaps most interestingly, it doesn't actually use a timer (a `time.Ticker`) to explicitly add tokens to buckets on some regular cadence. Rather, it refills buckets on demand, based on the time elapsed between requests. This strategy means that we don't have to dedicate background processes to filling buckets until they're actually used, which will scale much more effectively:

```
// Effector is the function that you want to subject to throttling.
type Effector func(context.Context) (string, error)

// Throttled wraps an Effector. It accepts the same parameters, plus a
// "UID" string that represents a caller identity. It returns the same,
// plus a bool that's true if the call is not throttled.
type Throttled func(context.Context, string) (bool, string, error)

// A bucket tracks the requests associated with a UID.
type bucket struct {
    tokens uint
    time   time.Time
}

// Throttle accepts an Effector function, and returns a Throttled
// function with a per-UID token bucket with a capacity of max
// that refills at a rate of refill tokens every d.
func Throttle(e Effector, max uint, refill uint, d time.Duration) Throttled {
    // buckets maps UIDs to specific buckets
    buckets := map[string]*bucket{}
```

9 Wikipedia contributors. "Token bucket." *Wikipedia, The Free Encyclopedia*, 5 Jun. 2019. *https://oreil.ly/vkOov*.

```
        return func(ctx context.Context, uid string) (bool, string, error) {
            b := buckets[uid]

            // This is a new entry! It passes. Assumes that capacity >= 1.
            if b == nil {
                buckets[uid] = &bucket{tokens: max - 1, time: time.Now()}

                str, err := e(ctx)
                return true, str, err
            }

            // Calculate how many tokens we now have based on the time
            // passed since the previous request.
            refillInterval := uint(time.Since(b.time) / d)
            tokensAdded := refill * refillInterval
            currentTokens := b.tokens + tokensAdded

            // We don't have enough tokens. Return false.
            if currentTokens < 1 {
                return false, "", nil
            }

            // If we've refilled our bucket, we can restart the clock.
            // Otherwise, we figure out when the most recent tokens were added.
            if currentTokens > max {
                b.time = time.Now()
                b.tokens = max - 1
            } else {
                deltaTokens := currentTokens - b.tokens
                deltaRefills := deltaTokens / refill
                deltaTime := time.Duration(deltaRefills) * d

                b.time = b.time.Add(deltaTime)
                b.tokens = currentTokens - 1
            }

            str, err := e(ctx)

            return true, str, err
        }
    }
```

Like the example in "Throttle" on page 86, this `Throttle` function accepts a function literal that conforms to the `Effector` contract, plus some values that define the size and refill rate of the underlying token bucket.

Instead of returning another `Effector`, however, it returns a `Throttled` function, which in addition to wrapping the effector with the throttling logic adds a "key" input parameter, which represents a unique user identifier, and a Boolean return value, which indicates whether the function has been throttled (and therefore not executed).

As interesting as you may (or may not) find the `Throttle` code, it's still not production ready. First of all, it's not entirely safe for concurrent use. A production implementation will probably want to lock on the `record` values, and possibly the `bucket` map. Second, there's no way to purge old records. In production, we'd probably want to use something like an LRU cache, like the one we described in "Efficient Caching Using an LRU Cache" on page 198, instead.

In the following, we show a toy example of how `Throttle` might be used in a RESTful web service:

```go
var throttled = Throttle(getHostname, 1, 1, time.Second)

func getHostname(ctx context.Context) (string, error) {
    if ctx.Err() != nil {
        return "", ctx.Err()
    }

    return os.Hostname()
}

func throttledHandler(w http.ResponseWriter, r *http.Request) {
    ok, hostname, err := throttled(r.Context(), r.RemoteAddr)

    if err != nil {
        http.Error(w, err.Error(), http.StatusInternalServerError)
        return
    }

    if !ok {
        http.Error(w, "Too many requests", http.StatusTooManyRequests)
        return
    }

    w.WriteHeader(http.StatusOK)
    w.Write([]byte(hostname))
}

func main() {
    r := mux.NewRouter()
    r.HandleFunc("/hostname", throttledHandler)
    log.Fatal(http.ListenAndServe(":8080", r))
}
```

The previous code creates a small web service with a single (somewhat contrived) endpoint at `/hostname` that returns the service's hostname. When the program is run, the `throttled` var is created by wrapping the `getHostname` function—which provides the actual service logic—by passing it to `Throttle`, which we defined previously.

When the router receives a request for the `/hostname` endpoint, the request is forwarded to the `throttledHandler` function, which performs the calls to `throttled`, receiving a `bool` indicating throttling status, the hostname `string`, and an `error` value. A defined error causes us to return a `500 Internal Server Error`, and a throttled request gets a `429 Too Many Requests`. If all else goes well, we return the hostname and a status `200 OK`.

Note that the bucket values are stored locally, so this implementation can't really be considered production-ready either. If you want this to scale out, you might want to store the record values in an external cache of some kind so that multiple service replicas can share them.

Load shedding

It's an unavoidable fact of life that, as load on a server increases beyond what it can handle, something eventually has to give.

Load shedding is a technique used to predict when a server is approaching that saturation point and then mitigating the saturation by dropping some proportion of traffic in a controlled fashion. Ideally, this will prevent the server from overloading and failing health checks, serving with high latency, or just collapsing in a graceless, uncontrolled failure.

Unlike quota-based throttling, load shedding is reactive, typically engaging in response to depletion of a resource like CPU, memory, or request-queue depth.

Perhaps the most straightforward form of load shedding is a per-task throttling that drops requests when one or more resources exceed a particular threshold. For example, if your service provides a RESTful endpoint, you might choose to to return an HTTP 503 (service unavailable). The `gorilla/mux` web toolkit, which we found very effective in Chapter 5 in the section "Building an HTTP Server with gorilla/mux" on page 114, makes this fairly straightforward by supporting "middleware" handler functions that are called on every request:

```go
const MaxQueueDepth = 1000

// Middleware function, which will be called for each request.
// If queue depth is exceeded, it returns HTTP 503 (service unavailable).
func loadSheddingMiddleware(next http.Handler) http.Handler {
    return http.HandlerFunc(func(w http.ResponseWriter, r *http.Request) {
        // CurrentQueueDepth is fictional and for example purposes only.
        if CurrentQueueDepth() > MaxQueueDepth {
            log.Println("load shedding engaged")

            http.Error(w,
                err.Error(),
                http.StatusServiceUnavailable)
            return
        }

        next.ServeHTTP(w, r)
    })
}

func main() {
    r := mux.NewRouter()

    // Register middleware
    r.Use(loadSheddingMiddleware)

    log.Fatal(http.ListenAndServe(":8080", r))
}
```

Gorilla Mux middlewares are called on every request, each taking a request, doing something with it, and passing it down to another middleware or the final handler. This makes them perfect for implementing general request logging, header manipulation, `ResponseWriter` hijacking, or in our case, resource-reactive load shedding.

Our middleware uses the fictional `CurrentQueueDepth()` (your actual function will depend on your implementation) to check the current queue depth, and rejects requests with an HTTP 503 (service unavailable) if the value is too high. More sophisticated implementations might even be smarter about choosing which work is dropped by prioritizing particularly important requests.

Graceful service degradation

Resource-sensitive load shedding works well, but in some applications it's possible to act a little more gracefully by significantly decreasing the quality of responses when the service is approaching overload. Such *graceful degradation* takes the concept of load shedding one step further by strategically reducing the amount of work needed to satisfy each request instead of just rejecting requests.

There are as many ways of doing this as there are services, and not every service can be degraded in a reasonable manner, but common approaches include falling back on cached data or less expensive—if less precise—algorithms.

Play It Again: Retrying Requests

When a request receives an error response, or doesn't receive a response at all, it should just try again, right? Well, kinda. Retrying makes sense, but it's a lot more nuanced than that.

Take this snippet for example, a version of which I've found in a production system:

```
res, err := SendRequest()
for err != nil {
    res, err = SendRequest()
}
```

It seems seductively straightforward, doesn't it? It *will* repeat failed requests, but that's also *exactly* what it will do. So when this logic was deployed to a few hundred servers and the service to which it was issuing requests went down, the entire system went with it. A review of the service metrics, shown in Figure 9-2, revealed this.

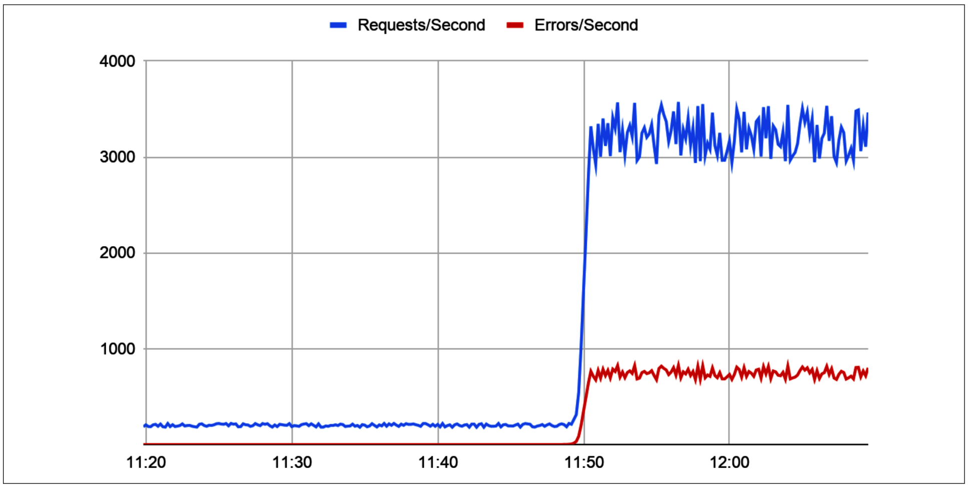


Figure 9-2. The anatomy of a "retry storm"

It seems that when the downstream service failed, our service—every single instance of it—entered its retry loop, making *thousands* of requests per second and bringing the network to its knees so severely that we were forced to essentially restart the entire system.

This is actually a very common kind of cascading failure known as a *retry storm*. In a retry storm, well-meaning logic intended to add resilience to a component, acts against the larger system. Very often, even when the conditions that caused the downstream service to go down are resolved, it can't come back up because it's instantly brought under too much load.

But, retries are a good thing, right?

Yes, but whenever you implement retry logic, you should always include a *backoff algorithm*, which we'll conveniently discuss in the next section.

Backoff Algorithms

When a request to a downstream service fails for any reason, "best" practice is to retry the request. But how long should you wait? If you wait too long, important work may be delayed. Too little and you risk overwhelming the target, the network, or both.

The common solution is to implement a backoff algorithm that introduces a delay between retries to reduce the frequency of attempts to a safe and acceptable rate.

There are a variety of backoff algorithms available, the simplest of which is to include a short, fixed-duration pause between retries, as follows:

```
res, err := SendRequest()
for err != nil {
    time.Sleep(2 * time.Second)
    res, err = SendRequest()
}
```

In the previous snippet, `SendRequest` is used to issue a request, returning string and error values. However, if `err` isn't `nil`, the code enters a loop, sleeping for two seconds before retrying, repeating indefinitely until it receives a nonerror response.

In Figure 9-3, we illustrate the number of requests generated by 1,000 simulated instances using this method.[10] As you can see, while the fixed-delay approach might reduce the request count compared to having no backoff at all, the overall number of requests is still quite consistently high.

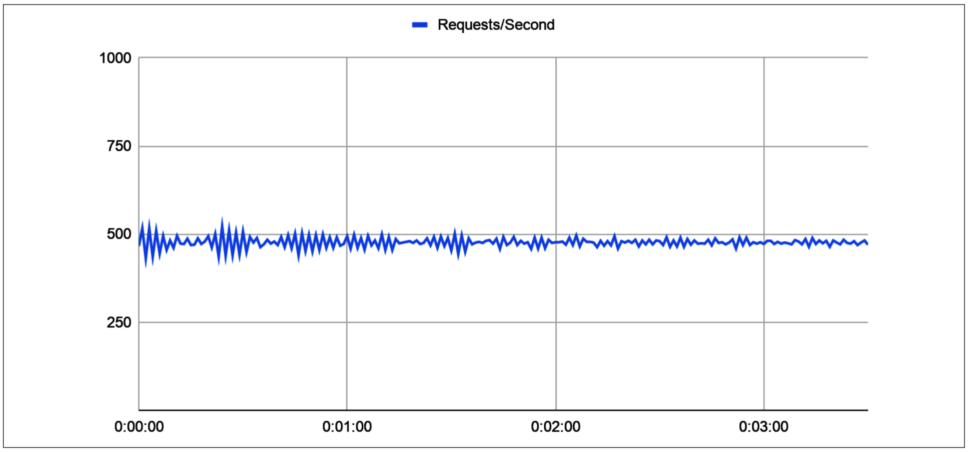


Figure 9-3. Requests/second of 1,000 simulated instances using a two-second retry delay

A fixed-duration backoff delay might work fine if you have a very small number of retrying instances, but it doesn't scale very well, since a sufficient number of requestors can still overwhelm the network.

However, we can't always assume that any given service will have a small enough number of instances not to overwhelm the network with retries, or that our service will even be the only one retrying. For this reason, many backoff algorithms

10 The code used to simulate all data in this section is available in the associated GitHub repository.

implement an *exponential backoff*, in which the durations of the delays between retries roughly doubles with each attempt up to some fixed maximum.

A very common (but flawed, as you'll soon see) exponential backoff implementation might look something like the following:

```
res, err := SendRequest()
base, cap := time.Second, time.Minute

for backoff := base; err != nil; backoff <<= 1 {
    if backoff > cap {
        backoff = cap
    }
    time.Sleep(backoff)
    res, err = SendRequest()
}
```

In this snippet, we specify a starting duration, `base`, and a fixed maximum duration, `cap`. In the loop, the value of `backoff` starts at `base` and doubles each iteration to a maximum value of `cap`.

You would think that this logic would help to mitigate the network load and retry request burden on downstream services. Simulating this implementation for 1,000 nodes, however, tells another story, illustrated in Figure 9-4.

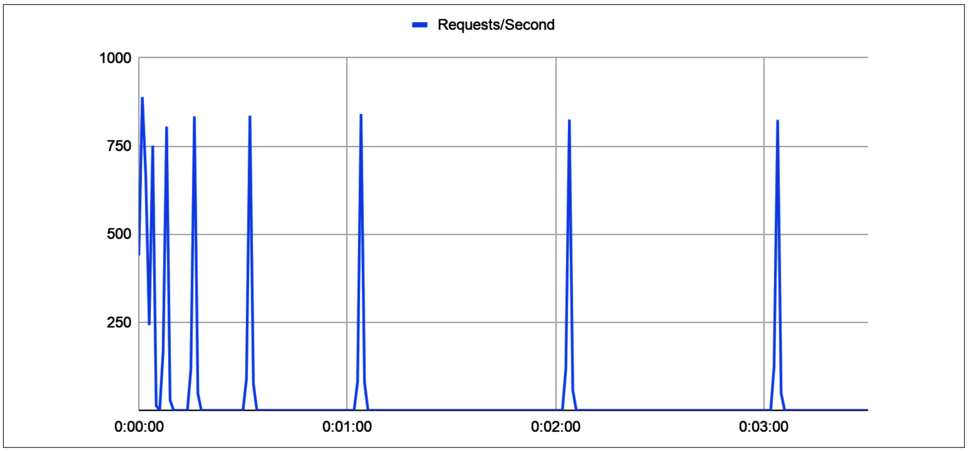


Figure 9-4. Requests/second of 1,000 simulated instances using an exponential backoff

It would seem that having 1,000 nodes with exactly the same retry schedule still isn't optimal, since the retries are now clustering, possibly generating enough load in the process to cause problems. So, in practice, pure exponential backoff doesn't necessarily help as much we'd like.

It would seem that we need some way to spread the spikes out so that the retries occur at a roughly constant rate. The solution is to include an element of randomness, called *jitter*. Adding jitter to our previous backoff function results in something like the snippet here:

```
res, err := SendRequest()
base, cap := time.Second, time.Minute

for backoff := base; err != nil; backoff <<= 1 {
    if backoff > cap {
        backoff = cap
    }

    jitter := rand.Int63n(int64(backoff * 3))
    sleep := base + time.Duration(jitter)
    time.Sleep(sleep)
    res, err = SendRequest()
}
```

Simulating running this code on 1,000 nodes produces the pattern presented in Figure 9-5.

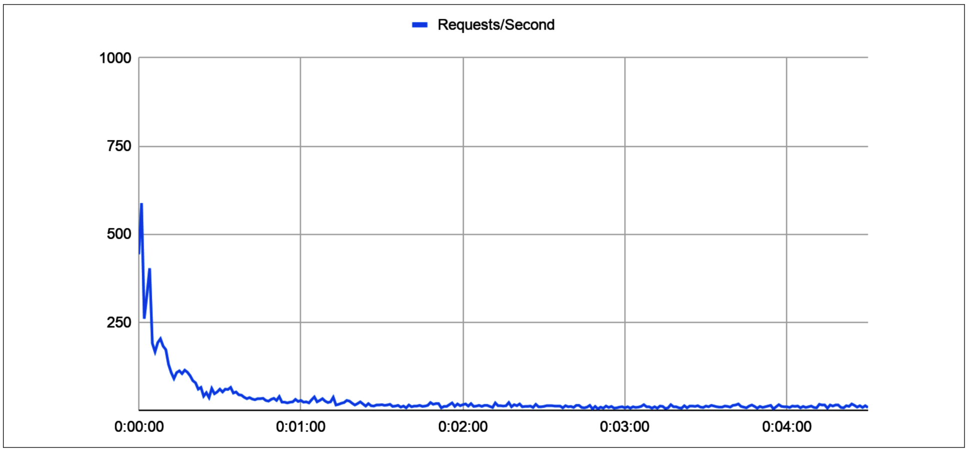


Figure 9-5. Requests/second of 1,000 simulated instances using an exponential backoff with jitter

The `rand` package's top-level functions produce a deterministic sequence of values each time the program is run. If you don't use the `rand.Seed` function to provide a new seed value, they behave as if seeded by `rand.Seed(1)` and always produce the same "random" sequence of numbers.

When we use exponential backoff with jitter, the number of retries decreases over a short interval—so as not to overstress services that are trying to come up—and spreads them out over time so that they occur at an approximately constant rate.

Who would have thought there was more to retrying requests than retrying requests?

Circuit Breaking

We first introduced the Circuit Breaker pattern in Chapter 4 as a function that degrades potentially failing method calls as a way to prevent larger or cascading failures. That definition still holds, and because we're not going to extend or change it much, we won't dig into it in *too* much detail here.

To review, the Circuit Breaker pattern tracks the number of consecutive failed requests made to a downstream component. If the failure count passes a certain threshold, the circuit is "opened," and all attempts to issue additional requests fail immediately (or return some defined fallback). After a waiting period, the circuit automatically "closes," resuming its normal state and allowing requests to be made normally.

Not all resilience patterns are defensive.

Sometimes it pays to be a good neighbor.

A properly applied Circuit Breaker pattern can make the difference between system recovery and cascading failure. In addition to the obvious benefits of not wasting resources or clogging the network with doomed requests, a circuit breaker (particularly one with a backoff function) can give a malfunctioning service enough room to recover, allowing it to come back up and restore correct service.

What's the Difference Between Circuit Breaker and Throttle?

At a quick glance, the Circuit Breaker pattern might seem to resemble a Throttle—after all they're both resilience patterns that rate requests—but they really are two quite different things:

- *Circuit Breaker* is generally applied only to *outgoing* requests. It usually doesn't care one bit about the request rate: it's only concerned with the number of failed requests, and only if they're consecutive.
- *Throttle* works like the throttle in a car by limiting a number of requests—regardless of success or failure—to some maximum rate. It's *typically* applied to incoming traffic, but there's no rule that says it has to be.

The Circuit Breaker pattern was covered in some detail in Chapter 4, so that's all we're going to say about it here. Take a look at "Circuit Breaker" on page 77 for more background and code examples. The addition of jitter to the example's backoff function is left as an exercise for the reader.[11]

Timeouts

The importance of timeouts isn't always appreciated. However, the ability for a client to recognize when a request is unlikely to be satisfied allows the client to release resources that it—and any upstream requestors it might be acting on behalf of—might otherwise hold on to. This holds just as true for a service, which may find itself holding onto requests until long after a client has given up.

For example, imagine a basic service that queries a database. If that database should suddenly slow so that queries take a few seconds to complete, requests to the service—each holding onto a database connection—could accumulate, eventually depleting the connection pool. If the database is shared, it could even cause other services to fail, resulting in a cascading failure.

If the service had timed out instead of holding on to the database, it could have degraded service instead of failing outright.

In other words, if you think you're going to fail, fail fast.

Using Context for service-side timeouts

We first introduced `context.Context` back in Chapter 4 as Go's idiomatic means of carrying deadlines and cancellation signals between processes.[12] If you'd like a refresher, or just want to put yourself in the right frame of mind before continuing, go ahead and take a look at "The Context Package" on page 72.

You might also recall that later in the same chapter, in "Timeout" on page 90, we covered the *Timeout* pattern, which uses `Context` to not only allow a process to stop waiting for an answer once it's clear that a result may not be coming, but to also notify other functions with derived `Contexts` to stop working and release any resources that they might also be holding on to.

This ability to cancel not just local functions, but subfunctions, is so powerful that it's generally considered good form for functions to accept a `Context` value if they have the potential to run longer than a caller might want to wait, which is almost always true if the call traverses a network.

11 Doing that here felt redundant, but I'll admit that I may have gotten a bit lazy.

12 And, technically, request-scoped values, but the correctness of this functionality is debatable.

For this reason, there are many excellent samples of Context-accepting functions scattered throughout Go's standard library. Many of these can be found in the sql package, which includes Context-accepting versions of many of its functions. For example, the DB struct's QueryRow method has an equivalent QueryRowContext that accepts a Context value.

A function that uses this technique to provide the username of a user based on an ID value might look something like the following:

```
func UserName(ctx context.Context, id int) (string, error) {
    const query = "SELECT username FROM users WHERE id=?"

    dctx, cancel := context.WithTimeout(ctx, 15*time.Second)
    defer cancel()

    var username string
    err := db.QueryRowContext(dctx, query, id).Scan(&username)

    return username, err
}
```

The UserName function accepts a context.Context and an id integer, but it also creates its own derived Context with a rather long timeout. This approach provides a default timeout that automatically releases any open connections after 15 seconds—longer than many clients are likely to be willing to wait—while also being responsive to cancellation signals from the caller.

The responsiveness to outside cancellation signals can be quite useful. The http framework provides yet another excellent example of this, as demonstrated in the following UserGetHandler HTTP handler function:

```
func UserGetHandler(w http.ResponseWriter, r *http.Request) {
    vars := mux.Vars(r)
    id := vars["id"]

    // Get the request's context. This context is canceled when
    // the client's connection closes, the request is canceled
    // (with HTTP/2), or when the ServeHTTP method returns.
    rctx := r.Context()

    ctx, cancel := context.WithTimeout(rctx, 10*time.Second)
    defer cancel()

    username, err := UserName(ctx, id)

    switch {
    case errors.Is(err, sql.ErrNoRows):
        http.Error(w, "no such user", http.StatusNotFound)
    case errors.Is(err, context.DeadlineExceeded):
        http.Error(w, "database timeout", http.StatusGatewayTimeout)
    case err != nil:
        http.Error(w, err.Error(), http.StatusInternalServerError)
    default:
        w.Write([]byte(username))
    }
}
```

In `UserGetHandler`, the first thing we do is retrieve the request's `Context` via its `Context` method. Conveniently, this `Context` is canceled when the client's connection closes, when the request is canceled (with HTTP/2), or when the `ServeHTTP` method returns.

From this we create a derived context, applying our own explicit timeout, which will cancel the `Context` after 10 seconds, no matter what.

Because the derived context is passed to the `UserName` function, we are able to draw a direct causative line between closing the HTTP request and closing the database connection: if the request's `Context` closes, all derived `Contexts` close as well, ultimately ensuring that all open resources are released as well in a loosely coupled manner.

Timing out HTTP/REST client calls

Back in "A Possible Pitfall of Convenience Functions" on page 229, we presented one of the pitfalls of the `http` "convenience functions" like `http.Get` and `http.Post`: that they use the default timeout. Unfortunately, the default timeout value is `0`, which Go interprets as "no timeout."

The mechanism we presented at the time for setting timeouts for client methods was to create a custom `Client` value with a nonzero `Timeout` value, as follows:

```
var client = &http.Client{
    Timeout: time.Second * 10,
}

response, err := client.Get(url)
```

This works perfectly fine, and in fact, will cancel a request in exactly the same way as if its `Context` is canceled. But what if you want to use an existing or derived `Context` value? For that you'll need access to the underlying `Context`, which you can get by using `http.NewRequestWithContext`, the `Context`-accepting equivalent of `http.NewRequest`, which allows a programmer to specify a `Context` that controls the entire lifetime of the request and its response.

This isn't as much of a divergence as it might seem. In fact, looking at the source code for the `Get` method on the `http.Client` shows that under the covers, it's just using `NewRequest`:

```
func (c *Client) Get(url string) (resp *Response, err error) {
    req, err := NewRequest("GET", url, nil)
```

```
        if err != nil {
            return nil, err
        }

        return c.Do(req)
    }
```

As you can see, the standard `Get` method calls `NewRequest` to create a `*Request` value, passing it the method name and URL (the last parameter accepts an optional `io.Reader` for the request body, which we don't need here). A call to the `Do` function executes the request proper.

Not counting an error check and the return, the entire method consists of just one call. It would seem that if we wanted to implement similar functionality that also accepts a `Context` value, we could do so without much hassle.

One way to do this might be to implement a `GetContext` function that accepts a `Context` value:

```
type ClientContext struct {
    http.Client
}

func (c *ClientContext) GetContext(ctx context.Context, url string)
        (resp *http.Response, err error) {

    req, err := http.NewRequestWithContext(ctx, "GET", url, nil)
    if err != nil {
        return nil, err
    }

    return c.Do(req)
}
```

Our new `GetContext` function is functionally identical to the canonical `Get`, except that it also accepts a `Context` value, which it uses to call `http.NewRequestWithContext` instead of `http.NewRequest`.

Using our new `ClientContext` would be very similar to using a standard `http.Client` value, except instead of calling `client.Get` we'd call `client.GetContext` (and pass along a `Context` value, of course):

```
func main() {
    client := &ClientContext{}
    ctx, cancel := context.WithTimeout(context.Background(), 5*time.Second)
    defer cancel()

    response, err := client.GetContext(ctx, "http://www.example.com")
    if err != nil {
        log.Fatal(err)
    }
    bytes, _ := ioutil.ReadAll(response.Body)
    fmt.Println(string(bytes))
}
```

But does it work? It's not a *proper* test with a testing library, but we can manually kick the tires by setting the deadline to `0` and running it:

```
$ go run .
2020/08/25 14:03:16 Get "http://www.example.com": context deadline exceeded
exit status 1
```

And it would seem that it does! Excellent.

Timing out gRPC client calls

Just like `http.Client`, gRPC clients default to "no timeout," but also allow timeouts to be explicitly set.

As we saw in "Implementing the gRPC client" on page 239, gRPC clients typically use the `grpc.Dial` function to establish a connection to a client, and that a list of `grpc.DialOption` values—constructed via functions like `grpc.WithInsecure` and `grpc.WithBlock`—can be passed to it to configure how that connection is set up.

Among these options is `grpc.WithTimeout`, which can be used to configure a client dialing timeout:

```
opts := []grpc.DialOption{
    grpc.WithInsecure(),
    grpc.WithBlock(),
    grpc.WithTimeout(5 * time.Second),
}
conn, err := grpc.Dial(serverAddr, opts...)
```

However, while `grpc.WithTimeout` might seem convenient on the face of it, it's actually been deprecated for some time, largely because its mechanism is inconsistent (and redundant) with the preferred `Context` timeout method. We show it here for the sake of completion.

The `grpc.WithTimeout` option is deprecated and will eventually be removed. Use `grpc.DialContext` and `context.WithTimeout` instead.

Instead, the preferred method of setting a gRPC dialing timeout is the very convenient (for us) `grpc.DialContext` function, which allows us to use (or reuse) a `con text.Context` value. This is actually doubly useful, because gRPC service methods accept a Context value anyway, so there really isn't even any additional work to be done:

```go
func TimeoutKeyValueGet() *pb.Response {
    // Use context to set a 5-second timeout.
    ctx, cancel := context.WithTimeout(context.Background(), 5 * time.Second)
    defer cancel()

    // We can still set other options as desired.
    opts := []grpc.DialOption{grpc.WithInsecure(), grpc.WithBlock()}

    conn, err := grpc.DialContext(ctx, serverAddr, opts...)
    if err != nil {
        grpclog.Fatalf(err)
    }
    defer conn.Close()

    client := pb.NewKeyValueClient(conn)

    // We can reuse the same Context in the client calls.
    response, err := client.Get(ctx, &pb.GetRequest{Key: key})
    if err != nil {
        grpclog.Fatalf(err)
    }

    return response
}
```

As advertised, TimeoutKeyValueGet uses grpc.DialContext—to which we pass a context.Context value with a 5-second timeout—instead of grpc.Dial. The opts list is otherwise identical except, obviously, that it no longer includes grpc.WithTimeout.

Note the client.Get method call. As we mentioned previously, gRPC service methods accept a Context parameter, so we simply reuse the existing one. Importantly, reusing the same Context value will constrain both operations under the same timeout calculation—a Context will time out regardless of how it's used—so be sure to take that into consideration when planning your timeout values.

Idempotence

As we discussed at the top of Chapter 4, cloud native applications by definition exist in and are subject to all of the idiosyncrasies of a networked world. It's a plain fact of life that networks—all networks—are unreliable, and messages sent across them don't always arrive at their destination on time (or at all).

What's more, if you send a message but don't get a response, you have no way to know what happened. Did the message get lost on its way to the recipient? Did the recipient get the message, and the response get lost? Maybe everything is working fine, but the round trip is just taking a little longer than usual?

In such a situation, the only option is to send the message again. But it's not enough to cross your fingers and hope for the best. It's important to plan for this inevitability by making it safe to resend messages by designing the functions for *idempotence*.

You might recall that we briefly introduced the concept of idempotence in "What Is Idempotence and Why Does It Matter?" on page 108, in which we defined an idempotent operation as one that has the same effect after multiple applications as a single application. As the designers of HTTP understood, it also happens to be an important property of any cloud native API that guarantees that any communication can be safely repeated (see "The Origins of Idempotence on the Web" on page 287 for a bit on that history).

The actual means of achieving idempotence will vary from service to service, but there are some consistent patterns that we'll review in the remainder of this section.

The Origins of Idempotence on the Web

The concepts of idempotence and safety, at least in the context of networked services, were first defined way back in 1997 in the HTTP/1.1 standard.[13]

An interesting aside: that ground-breaking proposal, as well as the HTTP/1.0 "informational draft" that preceded it the year before,[14] were authored by two greats.

The primary author of the original HTTP/1.0 draft (and the last author of the proposed HTTP/1.1 standard) was Sir Timothy John Berners-Lee, who is credited for inventing the World Wide Web, the first web browser, and the fundamental protocols and algorithms allowing the Web to scale—for which he was awarded with an ACM Turing Award, a knighthood, and various honorary degrees.

The primary author of the proposed HTTP/1.1 standard (and the second author of the original HTTP/1.0 draft) was Roy Fielding, then a graduate student at the University of California Irvine. Despite being one of the original authors of the World Wide Web, Fielding is perhaps known for his doctoral dissertation, in which he invented REST.[15]

13 Fielding, R., et al. "Hypertext Transfer Protocol — HTTP/1.1," Proposed Standard, RFC 2068, June 1997. *https://oreil.ly/28rcs*.

14 Berners-Lee, T., et al. "Hypertext Transfer Protocol — HTTP/1.0," Informational, RFC 1945, May 1996. *https://oreil.ly/zN7uo*.

15 Fielding, Roy Thomas. "Architectural Styles and the Design of Network-Based Software Architectures." *UC Irvine*, 2000, pp. 76–106. *https://oreil.ly/swjbd*.

How do I make my service idempotent?

Idempotence isn't baked into the logic of any particular framework. Even in HTTP—and by extension, REST—idempotence is a matter of convention and isn't explicitly enforced. There's nothing stopping you from—by oversight or on purpose—implementing a nonidempotent GET if you really want to.[16]

One of the reasons that idempotence is sometimes so tricky is because it relies on logic built into the core application, rather than at the REST or gRPC API layer. For example, if back in Chapter 5 we had wanted to make our key-value store consistent with traditional CRUD (create, read, update, and delete) operations (and therefore *not* idempotent) we might have done something like this:

```go
var store = make(map[string]string)

func Create(key, value string) error {
    if _, ok := store[key]; ok {
        return errors.New("duplicate key")
    }

    store[key] = value
    return nil
}

func Update(key, value string) error {
    if _, ok := store[key]; !ok {
        return errors.New("no such key")
    }

    store[key] = value
    return nil
}

func Delete(key string) error {
    if _, ok := store[key]; ok {
        return errors.New("no such key")
    }

    delete(store, key)
    return nil
}
```

This CRUD-like service implementation may be entirely well-meaning, but if any of these methods have to be repeated the result would be an error. What's more, there's also a fair amount of logic involved in checking against the current state which wouldn't be necessary in an equivalent idempotent implementation like the following:

16 You monster.

```
var store = make(map[string]string)

func Set(key, value string) {
    store[key] = value
}

func Delete(key string) {
    delete(store, key)
}
```

This version is a *lot* simpler, in more than one way. First, we no longer need separate "create" and "update" operations, so we can combine these into a single `Set` function. Also, not having to check the current state with each operation reduces the logic in each method, a benefit that continues to pay dividends as the service increases in complexity.

Finally, if an operation has to be repeated, it's no big deal. For both the `Set` and `Delete` functions, multiple identical calls will have the same result. They are idempotent.

What about scalar operations?

"So," you might say, "that's all well and good for operations that are either *done* or *not done*, but what about more complex operations? Operations on scalar values, for example?"

That's a fair question. After all, it's one thing to PUT a thing in a place: it's either been PUT, or it hasn't. All you have to do is not return an error for re-PUTs. Fine.

But what about an operation like "add $500 to account 12345"? Such a request might carry a JSON payload that looks something like the following:

```
{
    "credit":{
        "accountID": 12345,
        "amount": 500
    }
}
```

Repeated applications of this operation would lead to an extra $500 going to account 12345, and while the owner of the account might not mind so much, the bank probably would.

But consider what happens when we add a `transactionID` value to our JSON payload:

```
{
    "credit":{
        "accountID": 12345,
        "amount": 500,
        "transactionID": 789
        }
    }
```

It may require some more bookkeeping, but this approach provides a workable solution to our dilemma. By tracking `transactionID` values, the recipient can safely identify and reject duplicate transactions. Idempotence achieved!

Service Redundancy

Redundancy—the duplication of critical components or functions of a system with the intention of increasing reliability of the system—is often the first line of defense when it comes to increasing resilience in the face of failure.

We've already discussed one particular kind of redundancy—messaging redundancy, also known as "retries"—in "Play It Again: Retrying Requests" on page 275. In this section, however, we'll consider the value of replicating critical system components so that if any one fails, one or more others are there to pick up the slack.

In a public cloud, this would mean deploying your component to multiple server instances, ideally across multiple zones or even across multiple regions. In a container orchestration platform like Kubernetes, this may even just be a matter of setting your replica count to a value greater than one.

As interesting as this subject is, however, we won't actually spend too much time on it. Service replication is an architectural subject that's been thoroughly covered in many other sources.[17] This is supposed to be a Go book, after all. But still, we'd be remiss to have an entire chapter about resilience and not even mention it.

A Word of Caution: Fault Masking

Fault masking occurs when a system fault is invisibly compensated for without being explicitly detected.

For example, imagine a system with three service nodes, all performing a share of the tasks. If one node goes bad and the other nodes can compensate, you might never notice anything wrong. The fault has been masked.

Fault masking can conceal possibly progressive faults, and may eventually—and quietly—result in a loss of protective redundancy, often with a sudden and catastrophic outcome.

17 *Building Secure and Reliable Systems: Best Practices for Designing, Implementing, and Maintaining Systems* by Heather Adkins—and a host of other authors—is one excellent example.

To prevent fault masking, it's important to include service health checks—which we'll discuss in "Healthy Health Checks" on page 294—that accurately report the health of a service instance.

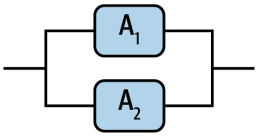
Designing for Redundancy

The effort involved in designing a system so that its functions can be replicated across multiple instances can yield significant dividends. But exactly how much? Well…a lot. You can feel free to take a look at the following box if you're interested in the math, but if you don't, you can just trust me on this one.

Reliability by the Numbers

Imagine, if you will, a service with "two-nines"—or 99%—of availability. Any given request to this system it has a theoretical probability of success of 0.99, which is denoted A_s. This actually isn't very good, but that's the point.

What kind of availability can you get if two of these identical instances are arranged in parallel, so that both have to be down to interrupt service?[18] This kind of arrangement can be diagrammed as shown here:



What's the resulting system availability? What we really want to know is: what's the probability that both instances will be *unavailable*? To answer this, we take the product of each component's probability of failure:

$$U_s = (1 - A_1) \times (1 - A_2)$$

This method generalizes to any number of components arranged in parallel, so that the availability of any N components is equal to one minus the product of their unavailabilities:

$$A_s = 1 - \prod_{1}^{N} (1 - A_i)$$

18 Brace yourself. We're going in.

When all A_i are equal, then this can be simplified to:

$$A_s = 1 - (1 - A_i)^N$$

So what about our example? Well, with two components, each with a 99% availability, we get the following:[19]

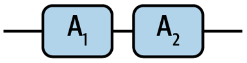
$$A_s = 1 - (1 - 0.99)^2 = 0.9999$$

99.99%. Four nines. That's an improvement of two orders of magnitude, which isn't half bad. But what if we added a third replica? Extending this out a little, we get some interesting results, summarized in the following table:

Components	Availability	Downtime per year	Downtime per month
One component	99% ("2-nines")	3.65 days	7.31 hours
Two parallel components	99.99% ("4-nines")	52.60 minutes	4.38 minutes
Three parallel components	99.9999% ("6-nines")	31.56 seconds	2.63 seconds

Incredibly, three parallel instances, each of which isn't exactly awesome on its own, can provide a very impressive 6-nines of availability! This is why cloud providers advise customers to deploy their applications with three replicas.

But what if the components are arranged serially, like a load balancer in front of our components? This might look something like the following:



In this kind of arrangement, if either component is unavailable, the entire system is unavailable. Its availability is the product of the availabilities of its components:

$$A_s = \prod_{1}^{N} A_i$$

When all A_i are equal, then this can be simplified to:

$$A_s = A^N$$

19 This assumes that the failure rates of the components are absolutely independent, which is very unlikely in the real world. Treat as you would spherical cows in a vacuum.

So what if we slapped a dodgy load balancer instance in front of our fancy 99.9999% available set of service replicas? As it turns out, the result isn't so good:

$$0.99 \times 0.999999 = 0.98999901$$

That's even lower than the load balancer by itself! This is important, because as it turns out:

The total reliability of a sequential system cannot be higher than the reliability of any one of its sequences of subsystems.

Autoscaling

Very often, the amount of load that a service is subjected to varies over time. The textbook example is the user-facing web service where load increases during the day and decreases at night. If such a service is built to handle the peak load, it's wasting time and money at night. If it's built only to handle the nighttime load, it will be overburdened in the daytime.

Autoscaling is a technique that builds on the idea of load balancing by automatically adding or removing resources—be they cloud server instances or Kubernetes pods—to dynamically adjust capacity to meet current demand. This ensures that your service can meet a variety of traffic patterns, anticipated or otherwise.

As an added bonus, applying autoscaling to your cluster can save money by right-sizing resources according to service requirements.

All major cloud providers provide a mechanism for scaling server instances, and most of their managed services implicitly or explicitly support autoscaling. Container orchestration platforms like Kubernetes also include support for autoscaling, both for the number of pods (horizontal autoscaling) and their CPU and memory limits (vertical autoscaling).

Autoscaling mechanics vary considerably between cloud providers and orchestration platforms, so a detailed discussion of how to gather metrics and configure things like predictive autoscaling is beyond the scope of this book. However, some key points to remember:

- Set reasonable maximums, so that unusually large spikes in demand (or, heaven forbid, cascade failures) don't completely blow your budget. The throttling and load shedding techniques that we discussed in "Preventing Overload" on page 270 are also useful here.
- Minimize startup times. If you're using server instances, bake machine images beforehand to minimize configuration time at startup. This is less of an issue on Kubernetes, but container images should still be kept small and startup times reasonably short.
- No matter how fast your startup, scaling takes a nonzero amount of time. Your service should have *some* wiggle room without having to scale.
- As we discussed in "Scaling Postponed: Efficiency" on page 197, the best kind of scaling is the kind that never needs to happen.

Healthy Health Checks

In "Service Redundancy" on page 290, we briefly discussed the value of redundancy—the duplication of critical components or functions of a system with the intention of increasing overall system reliability—and its value for improving the resilience of a system.

Multiple service instances means having a load-balancing mechanism—a service mesh or dedicated load balancer—but what happens when a service instance goes bad? Certainly, we don't want the load balancer to continue sending traffic its way. So what do we do?

Enter the *health check*. In its simplest and most common form, a health check is implemented as an API endpoint that clients—load balancers, as well as monitoring services, service registries, etc.—can use to ask a service instance if it's alive and healthy.

For example, a service might provide an HTTP endpoint (`/health` and `/healthz` are common naming choices) that returns a `200 OK` if the replica is healthy, and a `503 Service Unavailable` when it's not. More sophisticated implementations can even return different status codes for different states: HashiCorp's Consul service registry interprets any `2XX` status as a success, a `429 Too Many Requests` as a warning, and anything else as a failure.

Having an endpoint that can tell a client when a service instance is healthy (or not) sounds great and all, but it invites the question of what, exactly, does it mean for an instance to be "healthy"?

Health checks are like bloom filters. A failing health check means a service isn't up, but a health check passing means the service is *probably* "healthy." (Credit: Cindy Sridharan[20])

What Does It Mean for an Instance to Be "Healthy"?

We use the word "healthy" in the context of services and service instances, but what exactly do we mean when we say that? Well, as is so often the case, there's a simple answer and a complex answer. Probably a lot of answers in between, too.

We'll start with the simple answer. Reusing an existing definition, an instance is considered "healthy" when it's "available." That is, when it's able to provide correct service.

Unfortunately, it isn't always so clear cut. What if the instance itself is functioning as intended, but a downstream dependency is malfunctioning? Should a health check even make that distinction? If so, should the load balancer behave differently in each case? Should an instance be reaped and replaced if it's not the one at fault, particularly if all service replicas are affected?

Unfortunately, there aren't any easy answers to these questions, so instead of answers, I'll offer the next best thing: a discussion of the three most common approaches to health checking and their associated advantages and disadvantages. Your own implementations will depend on the needs of your service and your load-balancing behavior.

The Three Types of Health Checks

When a service instance fails, it's usually because of one of the following:

- A local failure like an application error or resource—CPU, memory, database connections, etc.—depletion.
- A remote failure in some dependency—a database or other downstream service—that affects the functioning of the service.

These two broad categories of failures give rise to three (yes, three) health checking strategies, each with its own fun little pros and cons.

Liveness checks do little more than return a "success" signal. They make no additional attempt to determine the status of the service, and say nothing about the service

20 Sridharan, Cindy (@copyconstruct). "Health checks are like bloom filters…" 5 Aug 2018, 3:21 AM. Tweet. *https://oreil.ly/Qpw3d*.

except that it's listening and reachable. But, then again, sometimes this is enough. We'll talk more about liveness checks in "Liveness checks" on page 296.

Shallow health checks go further than liveness checks by verifying that the service instance is likely to be able to function. These health checks only test local resources, so they're unlikely to fail on many instances simultaneously, but they can't say for certain whether a particular request service instance will be successful. We'll wade into shallow health checks in "Shallow health checks" on page 297.

Deep health checks provide a much better understanding of instance health, since they actually inspect the ability of a service instance to perform its function, which also exercises downstream resources like databases. While thorough, they can be expensive, and are susceptible to false positives. We'll dig into deep health checks in "Deep health checks" on page 298.

Liveness checks

A liveness endpoint always returns a "success" value, no matter what. While this might seem trivial to the point of uselessness—after all, what is the value of a health check that doesn't say anything about health—liveness probes actually can provide some useful information by confirming:

- That the service instance is listening and accepting new connections on the expected port
- That the instance is reachable over the network
- That any firewall, security group, or other configurations are correctly defined

This simplicity comes with a predictable cost, of course. The absence of any active health checking logic makes liveness checks of limited use when it comes to evaluating whether a service instance can actually perform its function.

Liveness probes are also dead easy to implement. Using the `net/http` package, we can do the following:

```
func healthLivenessHandler(w http.ResponseWriter, r *http.Request) {
    w.WriteHeader(http.StatusOK)
    w.Write([]byte("OK"))
})

func main() {
    r := mux.NewRouter()
    http.HandleFunc("/healthz", healthLivenessHandler)
    log.Fatal(http.ListenAndServe(":8080", r))
}
```

The previous snippet shows how little work can go into a liveness check. In it, we create and register a /healthz endpoint that does nothing but return a 200 OK (and the text OK, just to be thorough).

If you're using the gorilla/mux package, any registered middleware (like the load shedding function from "Load shedding" on page 274) can affect your health checks!

Shallow health checks

Shallow health checks go further than liveness checks by verifying that the service instance is *likely* to be able to function, but stop short of investigating in any way that might exercise a database or other downstream dependency.

Shallow health checks can evaluate any number of conditions that could adversely affect the service, including (but certainly not limited to):

- The availability of key local resources (memory, CPU, database connections)
- The ability to read or write local data, which checks disk space, permissions, and for hardware malfunctions such as disk failure
- The presence of support processes, like monitoring or updater processes

Shallow health checks are more definitive than liveness checks, and their specificity means that any failures are unlikely to affect the entire fleet at once.[21] However, shallow checks are prone to false positives: if your service is down because of some issue involving an external resource, a shallow check will miss it. What you gain in specificity, you also sacrifice in sensitivity.

A shallow health check might look something like the following example, which tests the service's ability to read and write to and from local disk:

```
func healthShallowHandler(w http.ResponseWriter, r *http.Request) {
    // Create our test file.
    // This will create a filename like /tmp/shallow-123456
    tmpFile, err := ioutil.TempFile(os.TempDir(), "shallow-")
    if err != nil {
        http.Error(w, err.Error(), http.StatusServiceUnavailable)
        return
    }
    defer os.Remove(tmpFile.Name())

    // Make sure that we can write to the file.
```

21 Though I've seen it happen.

```
        text := []byte("Check.")
        if _, err = tmpFile.Write(text); err != nil {
            http.Error(w, err.Error(), http.StatusServiceUnavailable)
            return
        }

        // Make sure that we can close the file.
        if err := tmpFile.Close(); err != nil {
            http.Error(w, err.Error(), http.StatusServiceUnavailable)
            return
        }

        w.WriteHeader(http.StatusOK)
    }

    func main() {
        r := mux.NewRouter()
        http.HandleFunc("/healthz", healthShallowHandler)
        log.Fatal(http.ListenAndServe(":8080", r))
    }
```

This simultaneously checks for available disk space, write permissions, and malfunctioning hardware, which can be a very useful thing to test, particularly if the service needs to write to an on-disk cache or other transient files.

An observant reader might notice that it writes to the default directory to use for temporary files. On Linux, this is `/tmp`, which is actually a RAM drive. This might be a useful thing to test as well, but if you want to test for the ability to write to disk on Linux you'll need to specify a different directory, or this becomes a very different test.

Deep health checks

Deep health checks directly inspect the ability of a service to interact with its adjacent systems. This provides much better understanding of instance health by potentially identifying issues with dependencies, like invalid credentials, the loss of connectivity to data stores, or other unexpected networking issues.

However, while thorough, deep health checks can be quite expensive. They can take a long time and place a burden on dependencies, particularly if you're running too many of them, or running them too often.

Don't try to test *every* dependency in your health checks: focus on the ones that are required for the service to operate.

When testing multiple downstream dependencies, evaluate them concurrently if possible.

What's more, because the failure of a dependency will be reported as a failure of the instance, deep checks are especially susceptible to false positives. Combined with the lower specificity compared to a shallow check—issues with dependencies will be felt by the entire fleet—and you have the potential for a cascading failure.

If you're using deep health checks, you should take advantage of strategies like circuit breaking (which we covered in "Circuit Breaking" on page 280) where you can, and your load balancer should "fail open" (which we'll discuss in "Failing Open" on page 300) whenever possible.

Here we have a trivial example of a possible deep health check that evaluates a database by calling a hypothetical service's `GetUser` function:

```go
func healthDeepHandler(w http.ResponseWriter, r *http.Request) {
    // Retrieve the context from the request and add a 5-second timeout
    ctx, cancel := context.WithTimeout(r.Context(), 5*time.Second)
    defer cancel()

    // service.GetUser is a hypothetical method on a service interface
    // that executes a database query
    if err := service.GetUser(ctx, 0); err != nil {
        http.Error(w, err.Error(), http.StatusServiceUnavailable)
        return
    }

    w.WriteHeader(http.StatusOK)
}

func main() {
    r := mux.NewRouter()
    http.HandleFunc("/healthz", healthDeepHandler)
    log.Fatal(http.ListenAndServe(":8080", r))
}
```

Ideally, a dependency test should execute an actual system function, but also be lightweight to the greatest reasonable degree. In this example, the `GetUser` function triggers a database query that satisfies both of these criteria.[22]

"Real" queries are generally preferable to just pinging the database for two reasons. First, they're a more representative test of what the service is doing. Second, they

22 It's an imaginary function, so let's just agree that that's true.

allow the leveraging of end-to-end query time as a measure of database health. The previous example actually does this—albeit in a very binary fashion—by using `Context` to set a hard timeout value, but you could choose to include more sophisticated logic instead.

Failing Open

What if all of your instances simultaneously decide that they're unhealthy? If you're using deep health checks, this can actually happen quite easily (and, perhaps, regularly). Depending on how your load balancer is configured, you might find yourself with zero instances serving traffic, possibly causing failures rippling across your system.

Fortunately, some load balancers handle this quite cleverly by "failing open." If a load balancer that fails open has *no* healthy targets—that is, if *all* of its targets' health checks are failing—it will route traffic to all of its targets.

This is slightly counterintuitive behavior, but it makes deep health checks somewhat safer to use by allowing traffic to continue to flow even when a downstream dependency may be having a bad day.

Summary

This was an interesting chapter to write. There's quite a lot to say about resilience, and so much crucial supporting operational background. I had to make some tough calls about what would make it in and what wouldn't. At about 37 pages, this chapter still turned out a fair bit longer than I intended, but I'm quite satisfied with the outcome. It's a reasonable compromise between too little information and too much, and between operational background and actual Go implementations.

We reviewed what it means for a system to fail, and how complex systems fail (that is, one component at a time). This led naturally to discussing a particularly nefarious, yet common, failure mode: cascading failures. In a cascade failure, a system's own attempts to recover hasten its collapse. We covered common measures of preventing cascading failures on the server side: throttling and load shedding.

Retries in the face of errors can contribute a lot to a service's resilience, but as we saw in the DynamoDB case study, can also contribute to cascade failures when applied naively. We dug deep into measures that can be taken on the client side as well, including circuit breakers, timeouts, and especially exponential backoff algorithms. There were several pretty graphs involved. I spent a lot of time on the graphs.

All of this led to conversations about service redundancy, how it affects reliability (with a little math thrown in, for fun), and when and how to best leverage autoscaling.

Of course, you can’t talk about autoscaling without talking about resource “health.” We asked (and did our best to answer) what it means for an instance to be “healthy,” and how that translated into health checks. We covered the three kinds of health checks and weighed their pros and cons, paying particular attention to their relative sensitivity/specificity tradeoffs.

In Chapter 10 we’ll take a break from the operational topics for a bit and wade into the subject of manageability: the art and science of changing the tires on a moving car.

CHAPTER 10
Manageability

Everyone knows that debugging is twice as hard as writing a program in the first place. So if you're as clever as you can be when you write it, how will you ever debug it?[1]

—Brian Kernighan, *The Elements of Programming Style (1978)*

In a perfect world, you'd never have to deploy a new version of your service or (heaven forbid!) shut down your entire system to fix or modify it to meet new requirements.

Then again, in a perfect world, unicorns would exist and four out of five dentists would recommend we eat pie for breakfast.[2]

Clearly, we don't live in a perfect world. But while unicorns might never exist,[3] you don't have to resign yourself to a world where you have to update your code whenever you need to alter your system's behavior.

While you'll probably always have to make code changes to update core logic, it is possible to build your systems so that you—or, critically, somebody else—can change a surprising variety of behaviors without having to recode and redeploy.

You may recall that we introduced this important attribute of cloud native systems back in "Manageability" on page 10, where we defined it as the ease with which a system's behavior can be modified to keep it secure, running smoothly, and compliant with changing requirements.

1 Kernighan, Brian W., and P. J. Plauger. *The Elements of Programming Style*. McGraw-Hill, 1978.

2 Staff, America's Test Kitchen. *Perfect Pie: Your Ultimate Guide to Classic and Modern Pies, Tarts, Galettes, and More*. America's Test Kitchen, 2019. *https://oreil.ly/rl5TP*.

3 They're doing some pretty amazing things with genetic engineering. Don't stop believing.

While this sounds straightforward, there's actually quite a bit more to manageability than you might think. It goes far beyond configuration files (though that's certainly part of it). In this chapter, we'll discuss what it means to have a manageable system, and we'll cover some of the techniques and implementations that can allow you to build a system that can change *almost* as quickly as its requirements.

Manageability Is Not Maintainability

It can be said that manageability and maintainability have some "mission overlap" in that they're both concerned with the ease with which a system can be modified. But where they differ somewhat is exactly *how* the system is modified:

- *Manageability* describes the ease with which changes can be made to the behavior of a system, typically without having to resort to changing its code. In other words, it's how easy it is to change to a system *from the outside*.
- *Maintainability* describes the ease with which a software system or component can be modified to change or add capabilities, correct faults or defects, or improve performance,[4] usually by making changes to the code. In other words, it's how easy it is to change a system *from the inside*.

What Is Manageability and Why Should I Care?

When considering manageability, it's common to think in terms of a single service. Can my service be configured easily? Does it have all the knobs and dials that it might need?

However, this misses the larger point by focusing on the component at the expense of the system. Manageability doesn't end at the service boundary. For a system to be manageable, the entire system has to be considered.

Take a moment to reconsider manageability with a complex system in mind. Can its behavior be easily modified? Can its components be modified independently of one another? Can they be easily replaced, if necessary? How do we know when that is?

Manageability encompasses all possible dimensions of a system's behavior. Its functions can be said to fall into four broad categories:[5]

4 "Systems and Software Engineering: Vocabulary." ISO/IEC/IEEE 24765:2010(E), 15 Dec. 2010. *https://oreil.ly/NInvC*.

5 Radle, Byron, et al. "What Is Manageability?" *NI*, National Instruments, 5 Mar. 2019. *https://oreil.ly/U3d7Q*.

Configuration and control

It's important that setting up and configuring a system—and each of its components—should be easily configurable for optimal availability and performance. Some systems need regular or real-time control, so having the right "knobs and levers" is absolutely fundamental. This is where we'll focus most of our attention in this chapter.

Monitoring, logging, and alerting

These functions keep track of the system's ability to do its job, and are critical to effective system management. After all, without them, how would we know when our system requires management? As vital as these features are to manageability, we won't discuss them in this chapter. Instead, they get an entire chapter of their own in Chapter 11, *Observability*.

Deployment and updates

Even in the absence of code changes, the ability to easily deploy, update, roll back, and scale system components is valuable, especially when there are many systems to manage. Obviously, this is useful during the initial deployment, but it comes into effect throughout a system's lifetime any time it has to be updated. Fortunately, its lack of external runtimes and singular executable artifacts make this an area in which Go excels.

Service discovery and inventory

A key feature of cloud native systems is their distributed nature. It's critical that components be able to quickly and accurately detect one another, a function called *service discovery*. Since service discovery is an architectural feature rather than a programmatic one, we won't go too deeply into it in this book.

Because this is more of a Go book than it is an architecture book,[6] it focuses largely on service implementations. For that reason only—*not because it's more important*—most of this chapter will similarly focus on service-level configuration. Unfortunately, an in-depth discussion of these is beyond the scope of this book.[7]

Managing complex computing systems is generally difficult and time-consuming, and the costs of managing them can far exceed the costs of the underlying hardware and software. By definition, a system designed to be manageable can be managed more efficiently, and therefore more cheaply. Even if you don't consider management costs, complexity reduction can have a huge impact on the likelihood of human error, making it easier and faster to undo when it inevitably creeps in. In that way,

6 Or so I told my editors. Hi, Amelia! Hi, Zan!

7 This makes me sad. These are important topics, but we have to focus.

manageability directly impacts reliability, availability, and security, making it a key ingredient of system dependability.

Configuring Your Application

The most basic function of manageability is the ability to configure an application. In an ideally configurable application, anything that's likely to vary between environments—staging, production, developer environments, etc.—will be cleanly separated from the code and be externally definable in some way.

You may recall that *The Twelve-Factor App*—a set of twelve rules and guidelines for building web applications that we introduced way back in Chapter 6—had quite a bit to say on this subject. In fact, the third of its twelve rules—"III. Configuration" on page 180—was concerned entirely with application configuration, about which it says:

> Store configuration in the environment.

As written, *The Twelve-Factor App* insists that all configurations should be stored in environment variables. There are a plenty of opinions on this, but in the years since its publication the industry seems to have reached a general consensus that what really matters is:

Configuration should be strictly separated from the code
: Configuration—anything that's likely to vary between environments—should always be cleanly separated from the code. While configuration can vary substantially across deploys, code does not. Configuration shouldn't be baked into the code. Ever.

Configurations should be stored in version control
: Storing configurations in version control—separately from the code—allows you to quickly roll back a configuration change if necessary, and aids system recreation and restoration. Some deployment frameworks, like Kubernetes, make this distinction naturally and relatively seamless by providing configuration primitives like the ConfigMap.

These days, it's still quite common to see applications configured mainly by environment variables, but it's just as common to see command-line flags and configuration files with various formats. Sometimes an application will even support more than one of these options. In the subsequent sections, we'll review some of these methods, their various pros and cons, and how they can be implemented in Go.

Configuration Good Practice

When you're building an application, you have a lot of options in how you define, implement, and deploy your application configurations. However, in my experience, I've found that certain general practices produce better long- and short-term outcomes:

Version control your configurations
Yes, I'm repeating myself, but this bears repeating. Configuration files should be stored in version control before being deployed to the system. This makes it possible to review them before deployment, to quickly reference them afterwards, and to quickly roll back a change if necessary. It's also helpful if (and when) you need to re-create and restore your system.

Don't roll your own format
Write your configuration files using a standard format like JSON, YAML, or TOML. We'll cover some of these later in the chapter. If you *must* roll your own format, be sure that you're comfortable with the idea of maintaining it—and forcing any future maintainers to deal with it—forever.

Make the zero value useful
Don't use nonzero default values unnecessarily. This is actually a good rule in general; there's even a "Go proverb" about it.[8] Whenever possible, the behavior that results from an undefined configuration should be acceptable, reasonable, and unsurprising. A simple, minimal configuration makes errors less likely.

Configuring with Environment Variables

As we discussed in Chapter 6, and reviewed previously, using environment variables to define configuration values is the method advocated for in *The Twelve-Factor App*. There's some merit to this preference: environment variables are universally supported, they ensure that configurations don't get accidentally checked into the code, and using them generally requires less code than using a configuration file. They're also perfectly adequate for small applications.

On the other hand, the process of setting and passing environment variables can be ugly, tedious, and verbose. While some applications support defining environment variables in a file, this largely defeats the purpose of using environment variables in the first place.

The implicit nature of environment variables can introduce some challenges as well. Since you can't easily learn about the existence and behavior of environment

8 Pike, Rob. "Go Proverbs." Gopherfest, 18 Nov. 2015, YouTube. *https://oreil.ly/5bOxW*.

variables by looking at an existing configuration file or checking the help output, applications that reply on them can sometimes be harder to use, and errors in them harder to debug.

As with most high-level languages, Go makes environment variables easily accessible. It does this through the standard `os` package, which provides the `os.Getenv` function for this purpose:

```
name := os.Getenv("NAME")
place := os.Getenv("CITY")

fmt.Printf("%s lives in %s.\n", name, place)
```

The `os.Getenv` function retrieves the value of the environment variable named by the key, but if the variable isn't present, it'll return an empty string. If you need to distinguish between an empty value and an unset value, Go also provides the `os.LookEnv` function, which returns both the value and a `bool` that's `false` if the variable isn't set:

```
if val, ok := os.LookupEnv(key); ok {
    fmt.Printf("%s=%s\n", key, val)
} else {
    fmt.Printf("%s not set\n", key)
}
```

This functionality is pretty minimal, but perfectly adequate for many (if not most) purposes. If you're in need of more sophisticated options, like default values or typed variables, there are several excellent third-party packages that provide this functionality. Viper (spf13/viper)—which we'll discuss in "Viper: The Swiss Army Knife of Configuration Packages" on page 329—is particularly popular.

Configuring with Command-Line Arguments

As configuration methods go, command-line arguments are definitely worth considering, at least for smaller, less-complex applications. After all, they're explicit, and details of their existence and usage are usually available via a `--help` option.

The standard flag package

Go includes the `flag` package, which is a basic command-line parsing package, in its standard library. While `flag` isn't particularly feature-rich, it's fairly straightforward to use, and—unlike `os.Getenv`—supports typing out-of-the-box.

Take, for example, the following program, which uses `flag` to implement a basic command that reads and outputs the values of command-line flags:

```
package main

import (
    "flag"
    "fmt"
)

func main() {
    // Declare a string flag with a default value "foo"
    // and a short description. It returns a string pointer.
    strp := flag.String("string", "foo", "a string")

    // Declare number and Boolean flags, similar to the string flag.
    intp := flag.Int("number", 42, "an integer")
    boolp := flag.Bool("boolean", false, "a boolean")

    // Call flag.Parse() to execute command-line parsing.
    flag.Parse()

    // Print the parsed options and trailing positional arguments.
    fmt.Println("string:", *strp)
    fmt.Println("integer:", *intp)
    fmt.Println("boolean:", *boolp)
    fmt.Println("args:", flag.Args())
}
```

As you can see from the previous code, the `flag` package allows you to register command-line flags with types, default values, and short descriptions, and to map those flags to variables. We can see a summary of these flags by running the program and passing it the `-help` flag:

```
$ go run . -help
Usage of /var/folders/go-build618108403/exe/main:
  -boolean
        a boolean
  -number int
        an integer (default 42)
  -string string
        a string (default "foo")
```

The help output presents us with a list of all of the available flags. Exercising all of these flags gives us something like the following:

```
$ go run . -boolean -number 27 -string "A string." Other things.
string: A string.
integer: 27
boolean: true
args: [Other things.]
```

It works! However, the `flag` package seems to have a couple of issues that limit its usefulness.

First, as you may have noticed, the resulting flag syntax seems a little…nonstandard. Many of us have come to expect command-line interfaces to follow the GNU argument standard, with long-named options prefixed by two dashes (`--version`) and short, single-letter equivalents (`-v`).

Second, all `flag` does is parse flags (though to be fair, it doesn't claim to do any more than that), and while that's nice, it's not as powerful as it could be. It sure would be nice if we could map commands to functions, wouldn't it?

The Cobra command-line parser

The `flags` package is perfectly fine if all you need to do is parse flags, but if you're in the market for something a little more powerful to build your command-line interfaces, you might want to consider the Cobra package. Cobra has a number of features that make it a popular choice for building fully featured command-line interfaces. It's used in a number of high-profile projects, including Kubernetes, CockroachDB, Docker, Istio, and Helm.

In addition to providing fully POSIX-compliant flags (short *and* long versions), Cobra also supports nested subcommands, and automatically generates help (`--help`) output and autocomplete for various shells. It also integrates with Viper, which we'll cover in "Viper: The Swiss Army Knife of Configuration Packages" on page 329.

Cobra's primary downside, as you might imagine, is that it's quite complex relative to the `flags` package. Using Cobra to implement the program from "The standard flag package" on page 308 looks like the following:

```go
package main

import (
    "fmt"
    "os"
    "github.com/spf13/cobra"
)

var strp string
var intp int
var boolp bool

var rootCmd = &cobra.Command{
    Use:  "flags",
    Long: "A simple flags experimentation command, built with Cobra.",
    Run:  flagsFunc,
}

func init() {
    rootCmd.Flags().StringVarP(&strp, "string", "s", "foo", "a string")
    rootCmd.Flags().IntVarP(&intp, "number", "n", 42, "an integer")
    rootCmd.Flags().BoolVarP(&boolp, "boolean", "b", false, "a boolean")
}

func flagsFunc(cmd *cobra.Command, args []string) {
    fmt.Println("string:", strp)
    fmt.Println("integer:", intp)
    fmt.Println("boolean:", boolp)
    fmt.Println("args:", args)
}

func main() {
    if err := rootCmd.Execute(); err != nil {
        fmt.Println(err)
        os.Exit(1)
    }
}
```

In contrast to the `flags`-package version, which basically just reads some flags and prints the results, the Cobra program has a bit more complexity, with several distinct parts.

First, we declare the target variables with package scope, rather than locally within a function. This is necessary because they have to be accessible to both the `init` function and the function that implements the command logic proper.

Next, we create a `cobra.Command` struct, `rootCmd`, that represents the *root command*. A separate `cobra.Command` instance is used to represent every command and subcommand that the CLI makes available. The `Use` field spells out the command's one-line usage message and `Long` is the long message displayed in the help output. `Run` is a function of type `func(cmd *Command, args []string)` that implements the actual work to be done when the command is executed.

Typically, commands are constructed in an `init` function. In our case, we add three flags—`string`, `number`, and `boolean`—to our root command along with their short flags, default values, and descriptions.

Every command gets an automatically generated help output, which we can retrieve using the `--help` flag:

```
$ go run . --help
A simple flags experimentation command, built with Cobra.

Usage:
  flags [flags]

Flags:
  -b, --boolean         a boolean
  -h, --help            help for flags
  -n, --number int      an integer (default 42)
  -s, --string string   a string (default "foo")
```

This makes sense, and it's also pretty! But does it run as we expect? Executing the command (using the standard flags style), gives us the following output:

```
$ go run . --boolean --number 27 --string "A string." Other things.
string: A string.
integer: 27
```

```
    boolean: true
    args: [Other things.]
```

The outputs are identical; we have achieved parity. But this is just a single command. One of the benefits of Cobra is that it also allows *subcommands*.

What does this mean? Take, for example, the `git` command. In this example, `git` would be the root command. By itself, it doesn't do much, but it has a series of subcommands—`git clone`, `git init`, `git blame`, etc.—that are related but are each distinct operations of their own.

Cobra provides this capability by treating commands as a tree structure. Each command and subcommand (including the root command) are represented by a distinct `cobra.Command` value. These are attached to one another using the `(c *Command) AddCommand(cmds ...*Command)` function. We demonstrate this in the following example by turning the `flags` command into a subcommand of a new root, which we call `cng` (for *Cloud Native Go*).

To do this, we first have to rename the original `rootCmd` to `flagsCmd`. We add a `Short` attribute to define its short description in help output, but it's otherwise identical. But now we need a new root command, so we create that as well:

```go
var flagsCmd = &cobra.Command{
    Use:   "flags",
    Short: "Experiment with flags",
    Long:  "A simple flags experimentation command, built with Cobra.",
    Run:   flagsFunc,
}

var rootCmd = &cobra.Command{
    Use:  "cng",
    Long: "A super simple command.",
}
```

Now we have two commands: the root command, `cng`, and a single subcommand, `flags`. The next step is to add the `flags` subcommand to the root command so that it's immediately beneath the root in the command tree. This is typically done in an `init` function, which we demonstrate here:

```go
func init() {
    flagsCmd.Flags().StringVarP(&strp, "string", "s", "foo", "a string")
    flagsCmd.Flags().IntVarP(&intp, "number", "n", 42, "an integer")
    flagsCmd.Flags().BoolVarP(&boolp, "boolean", "b", false, "a boolean")

    rootCmd.AddCommand(flagsCmd)
}
```

In the preceding `init` function, we keep the three `Flags` methods, except we now call them on `flagsCmd`.

What's new, however, is the `AddCommand` method, which allows us to add `flagsCmd` to `rootCmd` as a subcommand. We can repeat `AddCommand` as many times as we like with multiple `Command` values, adding as many subcommands (or sub-subcommands, or sub-sub-subcommands) as we want.

Now that we've told Cobra about the new `flags` subcommand, its information is reflected in the generated help output:

```
$ go run . --help
A super simple command.

Usage:
  cng [command]

Available Commands:
  flags       Experiment with flags
  help        Help about any command

Flags:
  -h, --help   help for cng

Use "cng [command] --help" for more information about a command.
```

Now, according to this help output, we have a top-level root command named `cng` that has *two* available subcommands: our `flags` command, and an automatically-generated `help` subcommand that lets a user view any subcommand's help. For example, `help flags` provides us with information and instructions for the `flags` subcommand:

```
$ go run . help flags
A simple flags experimentation command, built with Cobra.

Usage:
  cng flags [flags]

Flags:
  -b, --boolean         a boolean
  -h, --help            help for flags
  -n, --number int      an integer (default 42)
  -s, --string string   a string (default "foo")
```

Kind of neat, huh?

This is a tiny, tiny sample of what the Cobra library is capable of, but it's more than sufficient to let us to build a robust set of configuration options. If you're interested in learning more about Cobra and how you can use it to build powerful command-line interfaces, take a look at its GitHub repository and its listing on GoDoc.

Configuring with Files

Last but not least, we have what is probably the most commonly used configuration option: the configuration file.

Configuration files have a lot of advantages over environment variables, particularly for more complex applications. They tend to be more explicit and comprehensible by allowing behaviors to be logically grouped and annotated. Often, understanding how to use a configuration file is just a matter of looking at its structure or an example of its use.

Configuration files are particularly useful when managing a large number of options, which is an advantage they have over both environment variables and command-line flags. Command-line flags in particular can sometimes result in some pretty long statements that can be tedious and difficult to construct.

Files aren't the perfect solution though. Depending on your environment, distributing them at scale in a way that maintains parity across a cluster can be a challenge. This situation can be improved by having a single "source of truth," such as a distributed key/value store like etcd or HashiCorp Consul, or a central source code repository from which the deployment automatically draws its configuration, but this adds complexity and a dependency on another resource.

Fortunately, most orchestration platforms provide specialized configuration resources—such as Kubernetes' `ConfigMap` object—that largely alleviate the distribution problem.

There are probably dozens of file formats that have been used for configuration over the years, but in recent years, two in particular have stood out: JSON and YAML. In the next few sections, we'll go into each of these—and how to use them in Go—in a little more detail

Our configuration data structure

Before we proceed with a discussion of file formats and how to decode them, we should discuss the two general ways in which configurations can be unmarshalled:

- Configuration keys and values can be mapped to corresponding fields in a specific struct type. For example, a configuration that contains the attribute `host: localhost`, could be unmarshalled into a struct type that has a `Host string` field.
- Configuration data can be decoded and unmarshalled into one or more, possibly nested, maps of type `map[string]interface{}`. This can be convenient when you're working with arbitrary configurations, but it's awkward to work with.

If you know what your configuration is likely to look like in advance (which you generally do), then the first approach to decoding configurations, mapping them to a data structure created for that purpose, is by far the easiest. Although it's possible to decode and do useful work with arbitrary configuration schemas, doing so can be very tedious and isn't advisable for most configuration purposes.

So, for the remainder of this section, our example configurations will correspond to the following `Config` struct:

```go
type Config struct {
    Host string
    Port uint16
    Tags map[string]string
}
```

For a struct field to be marshallable or unmarshallable by *any* encoding package, it *must* begin with a capital letter to indicate that it's exported by its package.

For each of our examples we'll start with the `Config` struct, occasionally enhancing it with format-specific tags or other decorations.

Working with JSON

JSON (JavaScript Object Notation) was invented in the early 2000s, growing out of the need for a modern data interchange format to replace XML and other formats in use at the time. It's based on a subset of the JavaScript scripting language, making it both relatively human-readable and efficient for machines to generate and parse, while also offering the semantics for lists and mappings that were absent from XML.

As common and successful as JSON is, it does have some drawbacks. It's generally considered less user-friendly than YAML. Its syntax is especially unforgiving and can be easily broken by a misplaced (or missing) comma, and it doesn't even support comments.

However, of the formats presented in this chapter, it's the only one that's supported in Go's standard library.

What follows is a very brief introduction into encoding and decoding data to and from JSON. For a somewhat more thorough review, take a look at Andrew Gerrand's "JSON and Go" on *The Go Blog*.

Encoding JSON. The first step to understanding how to decode JSON (or any configuration format) is understanding how to *encode* it. This may seem strange, particularly in a section about reading configuration files, but encoding is important to the general subject of JSON encoding, and provides a handy means of generating, testing, and debugging your configuration files.[9]

JSON encoding and decoding is supported by Go's standard `encoding/json` package, which provides a variety of helper functions useful for encoding, decoding, formatting, validating, and otherwise working with JSON.

Among these is the `json.Marshal` function, which accepts an `interface{}` value `v`, and returns a `[]byte` array containing a JSON-encoded representation of `v`:

```go
func Marshal(v interface{}) ([]byte, error)
```

In other words, a value goes in, and JSON comes out.

This function really is as straightforward to use as it looks. For example, if we have an instance of `Config`, we can pass it to `json.Marshal` to get its JSON encoding:

```go
c := Config{
    Host: "localhost",
    Port: 1313,
    Tags: map[string]string{"env": "dev"},
}

bytes, err := json.Marshal(c)

fmt.Println(string(bytes))
```

If everything works as expected, `err` will be `nil`, and `bytes` will be a `[]byte` value containing the JSON. The `fmt.Println` output will look something like the following:

```
{"Host":"localhost","Port":1313,"Tags":{"env":"dev"}}
```

The `json.Marshal` function traverses the value of `v` recursively, so any internal structs will be encoded as well as nested JSON.

That was pretty painless, but if we're generating a configuration file it sure would be nice if the text was formatted for human consumption. Fortunately, `encoding/json`

9 Neat trick, huh?

also provides the following `json.MarshalIndent` function, which returns "pretty-printed" JSON:

```
func MarshalIndent(v interface{}, prefix, indent string) ([]byte, error)
```

As you can see, `json.MarshalIndent` works a lot like `json.Marshal`, except that also takes `prefix` and `indent` strings, as demonstrated here:

```
bytes, err := json.MarshalIndent(c, "", "   ")
fmt.Println(string(bytes))
```

The preceding snippet prints exactly what we'd hope to see:

```
{
   "Host": "localhost",
   "Port": 1313,
   "Tags": {
      "env": "dev"
   }
}
```

The result is prettily printed JSON, formatted for humans like you and me[10] to read. This is a very useful method for bootstrapping configuration files!

Decoding JSON. Now that we know how to encode a data structure into JSON, let's take a look at how to decode JSON as an existing data structure.

To do that, we use the conveniently named `json.Unmarshal` function:

```
func Unmarshal(data []byte, v interface{}) error
```

The `json.Unmarshal` function parses the JSON-encoded text contained in the `data` array and stores the result in the value pointed to by `v`. Importantly, if `v` is `nil` or isn't a pointer, `json.Unmarshal` returns an error.

But, what type should `v` be, exactly? Ideally, it would be a pointer to a data structure whose fields exactly correspond to the JSON structure. While it's possible to unmarshal arbitrary JSON into an unstructured map, as we'll discuss in "Decoding Arbitrary JSON" on page 318, this really should only be done if you really don't have any other choice.

As we'll see, though, if you have a data type that reflects your JSON's structure, then `json.Unmarshal` is able to update it directly. To do this, we first have to create an instance where our decoded data will be stored:

```
c := Config{}
```

10 Well, like you.

Now that we have our storage value, we can call `json.Unmarshal`, to which we pass a `[]byte` that contains our JSON data and a pointer to `c`:

```
bytes := []byte(`{"Host":"127.0.0.1","Port":1234,"Tags":{"foo":"bar"}}`)
err := json.Unmarshal(bytes, &c)
```

If `bytes` contains valid JSON, then `err` will be `nil` and the data from `bytes` will be stored in the struct `c`. Printing the value of `c` should now provide output like the following:

```
{127.0.0.1 1234 map[foo:bar]}
```

Neat! But what happens when the structure of the JSON doesn't exactly match the Go type? Let's find out:

```
c := Config{}
bytes := []byte(`{"Host":"127.0.0.1", "Food":"Pizza"}`)
err := json.Unmarshal(bytes, &c)
```

Interestingly, this snippet doesn't produce an error as you might expect. Instead, `c` now contains the following values:

```
{127.0.0.1 0 map[]}
```

It would seem that the value of `Host` was set, but `Food`, which has no corresponding value in the `Config` struct, was ignored. As it turns out, `json.Unmarshal` will only decode the fields that it can find in the target type. This behavior can actually be quite useful if you want to cherry pick a few specific fields out of a big JSON blob.

Decoding Arbitrary JSON

As we briefly mentioned in "Our configuration data structure" on page 314, it's *possible* to decode and do useful work with arbitrary JSON, but doing so can be pretty tedious and should only be done if you don't know the structure of your JSON beforehand.

Take, for example, this entirely arbitrary JSON data:

```
bytes := []byte(`{"Foo":"Bar", "Number":1313, "Tags":{"A":"B"}}`)
```

Without knowing this data's structure ahead of time, we can actually use `json.Unmarshal` to decode it into an `interface{}` value:

```
var f interface{}
err := json.Unmarshal(bytes, &f)
```

Outputting the new value of `f` with `fmt.Println` yields an interesting result:

```
map[Number:1313 Foo:Bar Tags:map[A:B]]
```

It would seem that the underlying value of `f` is now a map whose keys are strings and values are stored as empty interfaces. It's functionally identical to a value defined as:

```
    f := map[string]interface{}{
        "Foo":  "Bar",
        "Number":  1313,
        "Tags": map[string]interface{}{"A": "B"},
    }
```

But even though its underlying value is a `map[string]interface{}`, `f` still has a type of `interface{}`. We'll need to use a type assertion to access its values:

```
    m := f.(map[string]interface{})

    fmt.Printf("<%T> %v\n", m, m)
    fmt.Printf("<%T> %v\n", m["Foo"], m["Foo"])
    fmt.Printf("<%T> %v\n", m["Number"], m["Number"])
    fmt.Printf("<%T> %v\n", m["Tags"], m["Tags"])
```

Executing the previous snippet produces the following output:

```
    <map[string]interface {}> map[Number:1313 Foo:Bar Tags:map[A:B]]
    <string> Bar
    <float64> 1313
    <map[string]interface {}> map[A:B]
```

Field formatting with struct field tags. Under the covers, marshalling works by using reflection to examine a value and generate appropriate JSON for its type. For structs, the struct's field names are directly used as the default JSON keys, and the struct's field values become the JSON values. Unmarshalling works essentially the same way, except in reverse.

What happens when you marshal a zero-value struct? Well, as it turns out, when you marshal a `Config{}` value, for example, this is the JSON you get:

```
{"Host":"","Port":0,"Tags":null}
```

This isn't all that pretty. Or efficient. Is it really necessary to even output the empty values at all?

Similarly, struct fields have to be exported—and therefore capitalized—to be written or read. Does that mean that we're stuck with uppercase field names?

Fortunately, the answer to both questions is "no."

Go supports the use of *struct field tags*—short strings that appear in a struct after the type declaration of a field—that allow metadata to be added to specific struct fields. Field tags are most commonly used by encoding packages to modify encoding and decoding behavior at the field level.

Go struct field tags are special strings containing one or more keys/values pairs enclosed in backticks, after a field's type declaration:

```go
type User struct {
    Name string `example:"name"`
}
```

In this example, the struct's `Name` field is tagged with `example:"name"`. These tags can be accessed using run-time reflection via the `reflect` package, but their most common use case is to provide encoding and decoding directives.

The `encoding/json` package supports several such tags. The general format uses the `json` key in the struct field's tag, and a value that specifies the name of the field, possibly followed by a comma-separated list of options. The name may be empty in order to specify options without overriding the default field name.

The available options supported by `encoding/json` are shown here:

Customizing JSON keys
: By default, a struct field will case-sensitively map to a JSON key of the exact same name. A tag overrides this default name by setting the first (or only) value in the tag's options list.

  Example: `CustomKey string `json:"custom_key"``

Omitting empty values
: By default, a field will always appear in the JSON, even if it's empty. Using the `omitempty` option will cause fields to be skipped if they contain a zero-value. Note the leading comma in front of `omitempty`!

  Example: `OmitEmpty string `json:",omitempty"``

Ignoring a field
: Fields using the `-` (dash) option always will be completely ignored during encoding and decoding.

  Example: `IgnoredName string `json:"-"``

A struct that uses all of the previous tags might look like the following:

```go
type Tagged struct {
    // CustomKey will appear in JSON as the key "custom_key".
    CustomKey   string `json:"custom_key"`

    // OmitEmpty will appear in JSON as "OmitEmpty" (the default),
    // but will only be written if it contains a nonzero value.
    OmitEmpty   string `json:",omitempty"`

    // IgnoredName will always be ignored.
    IgnoredName string `json:"-"`

    // TwoThings will appear in JSON as the key "two_things",
    // but only if it isn't empty.
    TwoThings   string `json:"two_things,omitempty"`
}
```

For more information on how how `json.Marshal` encodes data, take a look at the function's documentation on golang.org.

Working with YAML

YAML (YAML Ain't Markup Language[11]) is an extensible file format that's popular with projects like Kubernetes that depend on complex, hierarchical configurations. It's highly expressive, though its syntax can also be a bit brittle, and configurations that use it can start to suffer from readability issues as they scale up.

Unlike JSON, which was originally created as a data interchange format, YAML is largely a configuration language at heart. Interestingly, however, YAML 1.2 is a superset of JSON, and the two formats are largely inter-convertible. YAML does have some advantages over JSON though: it can self-reference, it allows embedded block literals, and it supports comments and complex data types.

Unlike JSON, YAML isn't supported in Go's core libraries. While there are a few YAML packages to choose from, the standard choice is Go-YAML. Version 1 of Go-YAML started in 2014 as an internal project within Canonical to port the well-known `libyaml` C library to Go. As a project, it's exceptionally mature and well maintained. Its syntax is also conveniently very similar to `encoding/json`.

Encoding YAML. Using Go-YAML to encode data is a *lot* like encoding JSON. Exactly like it. In fact, the signatures for both packages' `Marshal` functions are identical. Like its `encoding/json` equivalent, Go-YAML's `yaml.Marshal` function also accepts an `interface{}` value, and returns its YAML encoding as a `[]byte` value:

```
func Marshal(v interface{}) ([]byte, error)
```

Just as we did in "Encoding JSON" on page 316, we demonstrate its use by creating an instance of `Config`, which we pass to `yaml.Marshal` to get its YAML encoding:

```
c := Config{
    Host: "localhost",
    Port: 1313,
    Tags: map[string]string{"env": "dev"},
}

bytes, err := yaml.Marshal(c)
```

11 Seriously, that really is what it stands for.

Once again, if everything works as expected, `err` will be `nil` and `bytes` will be a `[]byte` value containing the YAML. Printing the string value of `bytes` will provide something like the following:

```
host: localhost
port: 1313
tags:
  env: dev
```

Also, just like the version provided by `encoding/json`, Go-YAML's `Marshal` function traverses the value `v` recursively. Any composite types that it finds—arrays, slices, maps, and structs—will be encoded appropriately and will be present in the output as nested YAML elements.

Decoding YAML. In keeping with the theme we've established with the similarity of the `Marshal` functions from `encoding/json` and Go-YAML, the same consistency is evident between the two packages' `Unmarshal` functions:

```
func Unmarshal(data []byte, v interface{}) error
```

Again, the `yaml.Unmarshal` function parses the YAML-encoded data in the `data` array and stores the result in the value pointed to by `v`. If `v` is `nil` or not a pointer, `yaml.Unmarshal` returns an error. As shown here, the similarities are very clear:

```
// Caution: Indent this YAML with spaces, not tabs.
bytes := []byte(`
host: 127.0.0.1
port: 1234
tags:
    foo: bar
`)

c := Config{}
err := yaml.Unmarshal(bytes, &c)
```

Just as we did in "Decoding JSON" on page 317, we pass `yaml.Unmarshal` a pointer to a `Config` instance, whose fields correspond to the fields found in the YAML. Printing the value of `c` should (once again) provide output like the following:

```
{127.0.0.1 1234 map[foo:bar]}
```

There are other behavioral similarities between `encoding/json` and Go-YAML:

- Both will ignore attributes in a source document that cannot be mapped to the `Unmarshal` function. Again, this can be useful if you only care about a subset of the document, but it can be a "gotcha," too: if you forget to export the struct field, `Unmarshal` will always silently ignore it, and it'll never get set.
- Both are capable of unmarshalling arbitrary data by passing an `interface{}` value to `Unmarshal`. However, while `json.Unmarshal` will provide a

`map[string]interface{}`, `yaml.Unmarshal` will return a `map[interface{}]interface{}`. A minor difference, but another potential gotcha!

Struct field tags for YAML. In addition to the "standard" struct field tags—custom keys, `omitempty`, and `-` (dash)—detailed in "Field formatting with struct field tags" on page 319, Go-YAML supports two additional tags particular to YAML marshal formatting:

Flow style
: Fields using the `flow` option will be marshalled using the flow style, which can be useful for structs, sequences, and maps.

  Example: `Flow map[string]string `yaml:"flow"``

Inlining structs and maps
: The `inline` option causes all of a struct or map fields or keys to be processed as if they were part of the outer struct. For maps, keys must not conflict with the keys of other struct fields.

  Example: `Inline map[string]string `yaml:",inline"``

A struct that uses both of these options might look like the following:

```go
type TaggedMore struct {
    // Flow will be marshalled using a "flow" style
    // (useful for structs, sequences and maps).
    Flow map[string]string `yaml:"flow"`

    // Inlines a struct or a map, causing all of its fields
    // or keys to be processed as if they were part of the outer
    // struct. For maps, keys must not conflict with the yaml
    // keys of other struct fields.
    Inline map[string]string `yaml:",inline"`
}
```

As you can see, the tagging syntax is also consistent, except that instead of using the `json` prefix, Go-YAML tags use the `yaml` prefix.

Watching for configuration file changes

When working with configuration files, you'll inevitably be confronted with a situation in which changes have to be made to the configuration of a running program. If it doesn't explicitly watch for and reload changes, then it'll generally have to be restarted to reread its configuration, which can be inconvenient at best, and introduce downtime at worst.

At some point, you're going to have to decide how you want your program to respond to such changes.

The first (and least complex) option is to do nothing, and just expect the program to have to restart when its configuration changes. This is actually a fairly common choice, since it ensures that no trace of the former configuration exists. It also allows a program to "fail fast" when an error is introduced into the configuration file: the program just has to spit out an angry error message and refuse to start.

However, you might prefer to add logic to your program that detects changes in your configuration file (or files) and reloads them appropriately.

Making your configuration reloadable. If you'd like your internal configuration representations to reload whenever the underlying file changes, you'll have to plan a little ahead.

First, you'll want to have a single global instance of your configuration struct. For now, we'll use a `Config` instance of the kind we introduced in "Our configuration data structure" on page 314. In a slightly larger project, you might even put this in a `config` package:

```go
var config Config
```

Very often you'll see code in which an explicit `config` parameter is passed to just about every method and function. I've seen this quite a lot; often enough to know that this particular antipattern just makes life harder. Also, because the configuration now lives in N places instead of one, it also tends to make configuration reloading more complicated.

Once we have our `config` value, we'll want to add the logic that reads the configuration file and loads it into the struct. Something like the following `loadConfiguration` function will do just fine:

```go
func loadConfiguration(filepath string) (Config, error) {
    dat, err := ioutil.ReadFile(filepath)   // Ingest file as []byte
    if err != nil {
        return Config{}, err
    }

    config := Config{}

    err = yaml.Unmarshal(dat, &config)      // Do the unmarshal
    if err != nil {
        return Config{}, err
    }

    return config, nil
}
```

Our `loadConfiguration` function works almost the same way that we discussed in "Working with YAML" on page 321, except that it uses the `ioutil.ReadFile` function from the `io/ioutil` standard library to retrieve the bytes that it passes to yaml.Unmarshal. The choice to use YAML here was entirely arbitrary.[12] The syntax for a JSON configuration would be practically identical.

Now that we have logic to load our configuration file into a canonical struct, we need something to call it whenever it gets a notification that the file has changed. For that we have startListening, which monitors an updates channel:

```
func startListening(updates <-chan string, errors <-chan error) {
    for {
        select {
        case filepath := <-updates:
            c, err := loadConfiguration(filepath)
            if err != nil {
                log.Println("error loading config:", err)
                continue
            }
            config = c

        case err := <-errors:
            log.Println("error watching config:", err)
        }
    }
}
```

As you can see, startListening accepts two channels: updates, which emits the name of a file (presumably the configuration file) when that file changes, and an errors channel.

It watches both channels in a select inside of an infinite loop so that if a configuration file changes, the updates channel sends its name, which is then passed to loadConfiguration. If loadConfiguration doesn't return a non-nil error, then the Config value it returns replaces the current one.

Stepping back another level, we have an init function that retrieves the channels from a watchConfig function and passes them to startListening, which it runs as a goroutine:

```
func init() {
    updates, errors, err := watchConfig("config.yaml")
    if err != nil {
        panic(err)
    }

    go startListening(updates, errors)
}
```

12 Also, I just love JSON *so much*.

But what's this watchConfig function? Well, we don't quite know the details yet. We'll figure that out in the next couple of sections. We do know that it implements some configuration watching logic, and that it has a function signature that looks like the following:

```go
func watchConfig(filepath string) (<-chan string, <-chan error, error)
```

The watchConfig function, whatever its implementation, returns two channels—a string channel that sends the path of the updated configuration file, and an error channel that notifies about invalid configurations—and an error value that reports if there's a fatal error on startup.

The exact implementation of watchConfig can go a couple of different ways, each with its pros and cons. Now let's take a look at the two most common of those.

Polling for configuration changes. Polling, where you check for changes in your configuration file on some regular cadence, is a common way of watching a configuration file. A standard implementation uses a time.Ticker to recalculate a hash of your configuration file every few seconds and reload if the hash changes.

Go makes a number of common hash algorithms available in its crypto package, each of which lives in its own subpackage of crypto and satisfies both the crypto.Hash and io.Writer interfaces.

For example, Go's standard implementation of SHA256 can be found in crypto/sha256. To use it, you use its sha256.New function to get a new sha256.Hash value, into which you then write the data you want to calculate the hash of, just as you would any io.Writer. When that's complete, you use its Sum method to retrieve the resulting hash sum:

```go
func calculateFileHash(filepath string) (string, error) {
    file, err := os.Open(filepath)  // Open the file for reading
    if err != nil {
        return "", err
    }
    defer file.Close()              // Be sure to close your file!

    hash := sha256.New()            // Use the Hash in crypto/sha256

    if _, err := io.Copy(hash, file); err != nil {
        return "", err
    }

    sum := fmt.Sprintf("%x", hash.Sum(nil))  // Get encoded hash sum

    return sum, nil
}
```

Generating a hash for a configuration has three distinct parts. First, we get a []byte source in the form of an io.Reader. In this example we use an io.File. Next, we copy those bytes from the io.Reader to our sha256.Hash instance, which we do with a call to io.Copy. Finally, we use the Sum method to retrieve the hash sum from hash.

Now that we have our calculateFileHash function, creating our watchConfig implementation is just a matter of using a time.Ticker to concurrently check it on some cadence, and emit any positive results (or errors) to the appropriate channel:

```
func watchConfig(filepath string) (<-chan string, <-chan error, error) {
    errs := make(chan error)
    changes := make(chan string)
    hash := ""

    go func() {
        ticker := time.NewTicker(time.Second)

        for range ticker.C {
            newhash, err := calculateFileHash(filepath)
            if err != nil {
                errs <- err
                continue
            }

            if hash != newhash {
                hash = newhash
                changes <- filepath
            }
        }
    }()

    return changes, errs, nil
}
```

The polling approach has some benefits. It's not especially complex, which is always a big plus, and it works for any operating system. Perhaps most interestingly, because hashing only cares about the configuration's contents, it even can be generalized to detect changes in places like remote key/value stores that aren't technically files.

Unfortunately, the polling approach can be a little computationally wasteful, especially for very large or many files. By its nature, it also incurs also a brief delay between the time the file is changed and the detection of that change. If you're definitely working with local files, it would probably be more efficient to watch OS-level filesystem notifications, which we discuss in the next section.

Watching OS filesystem notifications. Polling for changes works well enough, but this method has some drawbacks. Depending on your use case, you may find it more efficient to instead monitor OS-level filesystem notifications.

Actually doing so, however, is complicated by the fact that each operating system has a different notification mechanism. Fortunately, the fsnotify package provides a workable abstraction that supports most operating systems.

To use this package to watch one or more files, you use the `fsnotify.NewWatcher` function to get a new `fsnotify.Watcher` instance, and use the `Add` method to register more files to watch. The `Watcher` provides two channels, `Events` and `Errors`, which sends notifications of file events and errors, respectively.

For example, if we wanted to watch our config file, we could do something like the following:

```go
func watchConfigNotify(filepath string) (<-chan string, <-chan error, error) {
    changes := make(chan string)

    watcher, err := fsnotify.NewWatcher()         // Get an fsnotify.Watcher
    if err != nil {
        return nil, nil, err
    }

    err = watcher.Add(filepath)                    // Tell watcher to watch
    if err != nil {                                // our config file
        return nil, nil, err
    }

    go func() {
        changes <- filepath                        // First is ALWAYS a change

        for event := range watcher.Events {        // Range over watcher events
            if event.Op&fsnotify.Write == fsnotify.Write {
                changes <- event.Name
            }
        }
    }()

    return changes, watcher.Errors, nil
}
```

Note the statement `event.Op & fsnotify.Write == fsnotify.Write`, which uses a bitwise AND (`&`) to filter for "write" events. We do this because the `fsnotify.Event` can potentially include multiple operations, each of which is represented as one bit in an unsigned integer. For example, a simultaneous `fsnotify.Write` (2, binary `0b00010`) and `fsnotify.Chmod` (16, binary `0b10000`) would result in an `event.Op` value of 18 (binary `0b10010`). Because `0b10010 & 0b00010 = 0b00010`, the bitwise AND allows us to guarantee that an operation includes a `fsnotify.Write`.

Viper: The Swiss Army Knife of Configuration Packages

Viper (spf13/viper) bills itself as a complete configuration solution for Go applications, and justifiably so. Among other things, it allows application configuration by a variety of mechanisms and formats, including, in order of precedence:

Explicitly set values
: This takes precedence over all other methods, and can be useful during testing.

Command-line flags
: Viper is designed to be a companion to Cobra, which we introduced in "The Cobra command-line parser" on page 310.

Environment variables
: Viper has full support for environment variables. Importantly, Viper treats environment variables as case-sensitive!

Configuration files, in multiple file formats
: Out of the box, Viper supports JSON and YAML with the packages we introduced previously; as well as TOML, HCL, INI, envfile, and Java Properties files. It can also write configuration files to help bootstrap your configurations, and even optionally supports live watching and rereading of configuration files.

Remote key/value stores
: Viper can access key/value stores like etcd or Consul, and can watch them for changes.

It also supports features like default values and typed variables, which the standard packages typically don't provide.

Keep in mind though, that while Viper does *a lot*, it's also a pretty big hammer that brings in a lot of dependencies. If you're trying to build a slim, streamlined application, Viper may be more than you need.

Explicitly setting values in Viper

Viper allows you to use the `viper.Set` function to explicitly set values from, for example, command-line flags or the application logic. This can be pretty handy during testing:

```
viper.Set("Verbose", true)
viper.Set("LogFile", LogFile)
```

Explicitly set values have the highest priority, and override values that would be set by other mechanisms.

Working with command-line flags in Viper

Viper was designed to be a companion to the Cobra library, which we briefly discussed in the context of constructing command-line interfaces in "The Cobra command-line parser" on page 310. This close integration with Cobra makes it straightforward to bind command-line flags to configuration keys.

Viper provides the `viper.BindPFlag` function, which allows individual command-line flags to be bound to a named key, and `viper.BindPFlags`, which binds a full flag set using each flag's long name as the key.

Because the actual value of the configuration value is set when the binding is accessed, rather than when it's called, you can call `viper.BindPFlag` in an `init` function as we do here:

```
var rootCmd = &cobra.Command{ /* omitted for brevity */ }

func init() {
    rootCmd.Flags().IntP("number", "n", 42, "an integer")
    viper.BindPFlag("number", rootCmd.Flags().Lookup("number"))
}
```

In the preceding snippet, we declare a `&cobra.Command` and define an integer flag called "number." Note that we use the `IntP` method instead of `IntVarP`, since there's no need to store the value of the flag in an external value when Cobra is used in this way. Then, using the `viper.BindPFlag` function, we bind the "number" flag to a configuration key of the same name.

After it's been bound (and the command-line flags parsed), the value of the bound key can be retrieved from Viper by using the `viper.GetInt` function:

```
n := viper.GetInt("number")
```

Working with environment variables in Viper

Viper provides several functions for working with environment variables as a configuration source. The first of these is `viper.BindEnv`, which is used to bind a configuration key to an environment variable:

```
viper.BindEnv("id")                     // Bind "id" to var "ID"
viper.BindEnv("port", "SERVICE_PORT")   // Bind "port" to var "SERVICE_PORT"

id := viper.GetInt("id")
id := viper.GetInt("port")
```

If only a key is provided, `viper.BindEnv` will bind to the environment variable matching the key. More arguments can be provided to specify one or more environment variables to bind to. In both cases, Viper automatically assumes that the name of the environment variable is in all caps.

Viper provides several additional helper functions for working with environment variables. See the Viper GoDoc for more details on these.

Working with configuration files in Viper

Out of the box, Viper supports JSON and YAML using the packages we introduced previously; as well as TOML, HCL, INI, envfile, and Java Properties files. It can also write configuration files to help bootstrap your configurations, and even optionally supports live watching and rereading of configuration files.

A discussion of local configuration files may seem unexpected in a book on cloud native, but files are still a commonly used structure in any context. After all, shared filesystems—be they Kubernetes ConfigMaps or NFS mounts—are quite common, and even cloud native services can be deployed by a configuration management system that installs a read-only local copy of a file for all service replicas to read. A configuration file could even be baked or mounted into a container image in a way that looks—as far as a containerized service is concerned—exactly like any other local file.

Reading configuration files. To read configurations from files, Viper just needs to know the names of the files, and where to look for them. Also, it needs to know its type, if that can't be inferred from a file extension. The `viper.ReadInConfig` function instructs Viper to find and read the configuration file, potentially returning an `error` value if something goes wrong. All of those steps are demonstrated here:

```go
viper.SetConfigName("config")

// Optional if the config has a file extension
viper.SetConfigType("yaml")

viper.AddConfigPath("/etc/service/")
viper.AddConfigPath("$HOME/.service")
viper.AddConfigPath(".")

if err := viper.ReadInConfig(); err != nil {
    panic(fmt.Errorf("fatal error reading config: %w", err))
}
```

As you can see, Viper can search multiple paths for a configuration file. Unfortunately, at this time, a single Viper instance only supports reading a single configuration file.

Watching and rereading configuration files in Viper. Viper natively allows your application to watch a configuration file for modifications and reload when changes are detected, which means that configurations can change without having to restart the server for them to take effect.

By default, this functionality is turned off. The `viper.WatchConfig` function can be used to enable it. Additionally, the `viper.OnConfigChange` function allows you to specify a function that's called whenever the configuration file is updated:

```go
viper.WatchConfig()
viper.OnConfigChange(func(e fsnotify.Event) {
    fmt.Println("Config file changed:", e.Name)
})
```

Make sure that any calls to `viper.AddConfigPath` are made *before* calling `viper.WatchConfig`.

Interestingly, Viper actually uses the `fsnotify/fsnotify` package behind the scenes, the same mechanism that we detailed in "Watching for configuration file changes" on page 323.

Using remote key/value stores with Viper

Perhaps the most interesting feature of Viper is its ability to read a configuration string written in any supported format from a path in a remote key/value store, like etcd or HashiCorp Consul. These values take precedence over default values, but are overridden by configuration values retrieved from disk, command-line flags, or environment variables.

To enable remote support in Viper, you first have to do a blank import of the `viper/remote` package:

```go
import _ "github.com/spf13/viper/remote"
```

A remote key/value configuration source can then be registered using the `viper.AddRemoteProvider` method, whose signature is as follows:

```go
func AddRemoteProvider(provider, endpoint, path string) error
```

- The `provider` parameter can be one of `etcd`, `consul`, or `firestore`.
- The `endpoint` is the URL of the remote resource. An odd quirk of Viper is that the etcd provider requires the URL to include a scheme (`http://ip:port`), while Consul requires *no scheme* (ip:port).
- The `path` is the path in the key-value store to retrieve the configuration from.

To read a JSON-formatted configuration file from an etcd service, for example, you'll do something like the following:

```
viper.AddRemoteProvider("etcd", "http://127.0.0.1:4001","/config/service.json")
viper.SetConfigType("json")
err := viper.ReadRemoteConfig()
```

Note that even though the configuration path includes a file extension, we also use `viper.SetConfigType` to explicitly define the configuration type. This is because from Viper's perspective, the resource is just a stream of bytes, so it can't automatically infer the format.[13] As of the time of writing, the supported formats are *json*, *toml*, *yaml*, *yml*, *properties*, *props*, *prop*, *env*, and *dotenv*.

Multiple providers may be added, in which case they're searched in the order in which they were added.

This is just a very basic introduction to what Viper can do with remote key/value stores. For more details about how to use Viper to read from Consul, watch for configuration changes, or read encrypted configurations, take a look at Viper's README.

Setting defaults in Viper

Unlike all of the other packages we reviewed in this chapter, Viper optionally allows default values to be defined for a key, by way of the `SetDefault` function.

Default values can sometimes be useful, but care should be taken with this functionality. As mentioned in "Configuration Good Practice" on page 307, useful zero values are generally preferable to implicit defaults, which can lead to surprising behaviors when thoughtlessly applied.

A snippet of Viper showing default values in action might look like the following:

```
viper.BindEnv("id")            // Will be upper-cased automatically
viper.SetDefault("id", "13")   // Default value is "13"

id1 := viper.GetInt("id")
fmt.Println(id1)               // 13

os.Setenv("ID", "50")          // Explicitly set the envvar

id2 := viper.GetInt("id")
fmt.Println(id2)               // 50
```

Default values have the lowest priority, and will only take effect if a key isn't explicitly set by another mechanism.

13 Or that feature just hasn't been implemented. I don't know.

Feature Management with Feature Flags

Feature flagging (or *feature toggling*[14]) is a software development pattern designed to increase the speed and safety with which new features can be developed and delivered by allowing specific functionality to be turned on or off during runtime, without having to deploy new code.

A feature flag is essentially a conditional in your code that enables or disables a feature based on some external criteria, often (but not always) a configuration setting. By setting the configuration to different values, a developer can, for example, choose to enable an incomplete feature for testing and disable it for other users.

Having the ability to release a product with unfinished features provides a number of powerful benefits.

First, feature flags allow many small incremental versions of software to be delivered without the overhead of branching and merging that comes with using feature branches. In other words, feature flags decouple the release of a feature from its deployment. Combined with the fact that feature flags, by their very nature, require code changes to be integrated as early as possible, which both encourages and facilitates continuous deployment and delivery. As a result, developers get more rapid feedback about their code, which in turn allows smaller, faster, and safer iterations.

Second, not only can feature flags allow features to be more easily tested before they're deemed ready for release, but they can also do so dynamically. For example, logic can be used to build feedback loops that can be combined with a circuit breaker–like pattern to enable or disable flags automatically under specific conditions.

Finally, logically executing flags can even be used to target feature rollouts to specific subsets of users. This technique, called *feature gating*, can be used as an alternative to proxy rules for canary deployments and staged or geographically based rollouts. When combined with observability techniques, feature gating can even allow you to more easily execute experiments like A/B testing or targeted tracing that instrument particular slices of user base, or even single customers.

The Evolution of a Feature Flag

In this section, we'll step through the iterative implementation of a feature flag with a function taken directly from the key-value REST service that we built in Chapter 5. Starting with the baseline function, we'll progress through several evolutionary

14 I've also seen "feature switch," "feature flipper," "conditional feature," and more. The industry seems to be settling on "flag" and "toggle," probably because the other names are just a little silly.

stages, from flaglessness all the way to a dynamic feature flag that toggles on for a particular subset of users.

In our scenario, we've decided that we want to be able to scale our key-value store, so we want to update the logic so that it's backed by a fancy distributed data structure instead of a local map.

Generation 0: The Initial Implementation

For our first iteration, we'll start with the `keyValueGetHandler` function from "Implementing the read function" on page 120. You may recall that `keyValueGetHandler` is an HTTP handler function that satisfies the `HandlerFunc` interface defined in the `net/http` package. If you're a little rusty on what that means, you may want to take a look back at "Building an HTTP Server with net/http" on page 112.

The initial handler function, copied almost directly from Chapter 5 (minus some of its error handling, for brevity) is shown here:

```go
func keyValueGetHandler(w http.ResponseWriter, r *http.Request) {
    vars := mux.Vars(r)                         // Retrieve "key" from the request
    key := vars["key"]

    value, err := Get(key)                      // Get value for key
    if err != nil {                             // Unexpected error!
        http.Error(w,
            err.Error(),
            http.StatusInternalServerError)
        return
    }

    w.Write([]byte(value))                      // Write the value to the response
}
```

As you can see, this function has no feature toggle logic (or indeed anything to toggle *to*). All it does is retrieve the key from the request variables, use the `Get` function to retrieve the value associated with that key, and write that value to the response.

In our next implementation, we'll start testing a new feature: a fancy distributed data structure to replace the local `map[string]string` that'll allow the service to scale beyond a single instance.

Generation 1: The Hard-Coded Feature Flag

In this implementation, we'll imagine that we've built our new and experimental distributed backend, and made it accessible via the `NewGet` function.

Our first attempt at creating a feature flag introduces a condition that allows us to use a simple Boolean value, `useNewStorage`, to switch between the two implementations:

```
// Set to true if you're working on the new storage backend
const useNewStorage bool = false;

func keyValueGetHandler(w http.ResponseWriter, r *http.Request) {
    vars := mux.Vars(r)
    key := vars["key"]

    var value string
    var err error

    if useNewStorage {
        value, err = NewGet(key)
    } else {
        value, err = Get(key)
    }

    if err != nil {
        http.Error(w,
            err.Error(),
            http.StatusInternalServerError)
        return
    }

    w.Write([]byte(value))
}
```

This first iteration shows some progress, but it's far from where we want to be. Having the flag condition fixed in the code as a hard-coded value makes it possible to toggle between implementations well enough for local testing, but it won't be easy to test both together in an automated and continuous manner.

Plus, you'll have to rebuild and redeploy the service whenever you want to change the algorithm you're using in a deployed instance, which largely negates the benefits of having a feature flag in the first place.

Practice good feature flag hygiene! If you haven't updated a feature flag in a while, consider removing it.

Generation 2: The Configurable Flag

A little time has gone by, and the shortcomings of hard-coded feature flags have become evident. For one thing, it would be really nice if we could use an external mechanism to change the value of the flag so we can test *both* algorithms in our tests.

In this example, we use Viper to bind and read an environment variable, which we can now use to enable or disable the feature at runtime. The choice of configuration mechanism isn't really important here. All that matters is that we're able to externally update the flag without having to rebuild the code:

```go
func keyValueGetHandler(w http.ResponseWriter, r *http.Request) {
    vars := mux.Vars(r)
    key := vars["key"]

    var value string
    var err error

    if FeatureEnabled("use-new-storage", r) {
        value, err = NewGet(key)
    } else {
        value, err = Get(key)
    }

    if err != nil {
        http.Error(w,
            err.Error(),
            http.StatusInternalServerError)
        return
    }

    w.Write([]byte(value))
}

func FeatureEnabled(flag string, r *http.Request) bool {
    return viper.GetBool(flag)
}
```

In addition to using Viper to read the environment variable that sets the `use-new-storage` flag, we've also introduced a new function: `FeatureEnabled`. At the moment all this does is perform `viper.GetBool(flag)`, but more importantly it also concentrates the flag reading logic in a single place. We'll see exactly what the benefit of this is in the next iteration.

You might be wondering why `FeatureEnabled` accepts an `*http.Request`. Well, it doesn't use it yet, but it'll make sense in the next iteration.

Generation 3: Dynamic Feature Flags

The feature is now deployed, but turned off behind a feature flag. Now we'd like to be able to test it in production on a specific subset of your user base. It's clear that we're not going to be able to implement this kind of flag with a configuration setting. Instead, we'll have to build dynamic flags that can *figure out for themselves* if they should be set. That means associating flags with functions.

Dynamic flags as functions

The first step in building dynamic flag functions is deciding what the signature of the functions will be. While it's not strictly required, it's helpful to define this explicitly with a function type like the one shown here:

```go
type Enabled func(flag string, r *http.Request) (bool, error)
```

The `Enabled` function type is the prototype for all of our dynamic feature flags functions. Its contract defines a function that accepts the flag name as a `string` and the `*http.Request`, and it returns a `bool` that's `true` if the requested flag is enabled.

Implementing a dynamic flag function

Using the contract provided by the `Enabled` type, we can now implement a function that we can use to determine whether a request is coming from a private network by comparing the request's remote address against a standard list of IP ranges allocated for private networks:

```go
// The list of CIDR ranges associated with internal networks.
var privateCIDRs []*net.IPNet

// We use an init function to load the privateCIDRs slice.
func init() {
    for _, cidr := range []string{
        "10.0.0.0/8",
        "172.16.0.0/12",
        "192.168.0.0/16",
    } {
        _, block, _ := net.ParseCIDR(cidr)
        privateCIDRs = append(privateCIDRs, block)
    }
}

// fromPrivateIP receives the flag name (which it ignores) and the
// request. If the request's remote IP is in a private range per
// RFC1918, it returns true.
func fromPrivateIP(flag string, r *http.Request) (bool, error) {
    // Grab the host portion of the request's remote address
    remoteIP, _, err := net.SplitHostPort(r.RemoteAddr)
    if err != nil {
        return false, err
    }

    // Turn the remote address string into a *net.IPNet
    ip := net.ParseIP(remoteIP)
    if ip == nil {
        return false, errors.New("couldn't parse ip")
    }

    // Loopbacks are considered "private."
    if ip.IsLoopback() {
        return true, nil
    }

    // Search the CIDRs list for the IP; return true if found.
    for _, block := range privateCIDRs {
        if block.Contains(ip) {
            return true, nil
        }
    }

    return false, nil
}
```

As you can see, the `fromPrivateIP` function conforms to `Enabled` by receiving a `string` value (the flag name) and an `*http.Request` (specifically, the instance associated with the initiating request). It returns `true` if the request originates from a private IP range (as defined by RFC 1918).

To make this determination, the `fromPrivateIP` function first retrieves the remote address, which contains the network address that sent the request, from the `*http.request`. After parsing off the host IP with `net.SplitHostPort` and using `net.ParseIP` to parse it into a `*net.IP` value, it compares the originating IP against each of the private CIDR ranges contained in `privateCIDRs`, returning `true` if a match is found.

This function also returns `true` if the request is traversing a load balancer or reverse proxy. A production-grade implementation will need to be aware of this, and would ideally be proxy protocol-aware.

Of course, this function is just an example. I used it because it's relatively simple, but a similar technique can be used to enable or disable a flag for a geographic region, a fixed percentage of users, or even for a specific customer.

The flag function lookup

Now that we have a dynamic flag function in the form of `fromPrivateIP`, we have to implement some mechanism of associating flags with it, by name. Perhaps the most straightforward way of doing this is to use a map of flag name strings to `Enabled` functions:

```
var enabledFunctions map[string]Enabled

func init() {
    enabledFunctions = map[string]Enabled{}
```

```
        enabledFunctions["use-new-storage"] = fromPrivateIP
    }
```

Using a map in this manner to indirectly reference functions provides us with a good deal of flexibility. We can even associate a function with multiple flags, if we like. This could be useful if we want a set of related features to always be active under the same conditions.

You may have noticed that we're using an `init` function to fill the `enabledFunctions` map. But wait, didn't we already have an `init` function?

Yes, we did, and that's okay. The `init` function is special: you're allowed to have multiple `init` functions if you like.

The router function

Finally, we get to tie everything together.

We do this by refactoring the `FeatureEnabled` function to look up the appropriate dynamic flag function, call it if it finds it, and return the result:

```
    func FeatureEnabled(flag string, r *http.Request) bool {
        // Explicit flags take precedence
        if viper.IsSet(flag) {
            return viper.GetBool(flag)
        }

        // Retrieve the flag function, if any. If none exists,
        // return false
        enabledFunc, exists := enabledFunctions[flag]
        if !exists {
            return false
        }

        // We now have the flag function: call it and return
        // the result
        result, err := enabledFunc(flag, r)
        if err != nil {
            log.Println(err)
            return false
        }

        return result
    }
```

At this point, `FeatureEnabled` has become a full-fledged router function that can dynamically control which code path is live according to explicit feature-flag settings and the output of flag functions. In this implementation, flags that have been explicitly set take precedence over everything else. This allows automated tests to verify both sides of a flagged feature.

Our implementation uses a simple in-memory lookup to determine the behavior of particular flags, but this could just as easily be implemented with a database or other data source, or even a sophisticated managed service like LaunchDarkly. Keep in mind, though, that these solutions do introduce a new dependency.

Feature Flags as a Service?

If you're interested in implementing some of the more sophisticated dynamic feature flags but (probably wisely) would prefer not to roll your own, LaunchDarkly provides an excellent "feature flags as a service" service.

Summary

Manageability isn't the most glamorous subject in the cloud native world—or any world, really—but I still really enjoyed how much we got our hands dirty with details in this chapter.

We dug into some of the nuts and bolts of various configuration styles, including environment variables, command-line flags, and variously formatted files. We even went over a couple of strategies for detecting configuration changes to trigger a reload. That's not to mention Viper, which pretty much does all of that and more.

I do feel like there may be some potential to go a lot deeper on some things, and I might have had it not been for the constraints of time and space. Feature flags and feature management are a pretty big subject, for example, and I definitely would have liked to have been able to explore them a bit more. Some subjects, like deployments and service discovery, we couldn't even cover at all. I guess we have some things to look forward to in the next edition, right?

As much as I enjoyed this chapter, I'm especially excited about Chapter 11, in which we'll get to dive into observability in general, and OpenTelemetry in particular.

Finally, I'll leave you with some advice: always be yourself, and remember that luck comes from hard work.

CHAPTER 11

Observability

Data is not information, information is not knowledge, knowledge is not understanding, understanding is not wisdom.[1]

—Clifford Stoll, *High-Tech Heretic: Reflections of a Computer Contrarian*

"Cloud native" is still a pretty new concept, even for computing. As far as I can tell, the term "cloud native" only started entering our vocabulary just after the founding of the Cloud Native Computing Foundation in the middle of 2015.[2]

As an industry, we're largely still trying to figure out exactly what "cloud native" means, and with each of the major public cloud providers regularly launching new services—each seeming to offer more abstraction than the last—even what little agreement we have is shifting over time.

One thing is clear, though: the functions (and failures) of the network and hardware layers are being increasingly abstracted and replaced with API calls and events. Every day we move closer to a world of software-defined *everything*. All of our problems are becoming software problems.

While we certainly sacrifice a fair share of control over the platforms our software runs on, we win *big* in overall manageability and reliability,[3] allowing us to focus our limited time and attention on our software. However, this also means that most of our failures now originate from within our own services and the interactions between them. No amount of fancy frameworks or protocols can solve the problem of bad

1 Stoll, Clifford. *High-Tech Heretic: Reflections of a Computer Contrarian*. Random House, September 2000.

2 Interestingly, this was also just after the AWS launched its Lambda functions as a service (FaaS) offering. Coincidence? Maybe.

3 Assuming of course that all of our network and platform configurations are correct!

software. As I said way back in Chapter 1, a kludgy application in Kubernetes is still kludgy.

Things are complicated in this brave new software-defined, highly distributed world. The software is complicated, the platforms are complicated, together they're *really* complicated, and more often than not we have no idea what's going on. Gaining visibility into our services has become more important than ever, and about the only thing that we *do* know is that the existing monitoring tools and techniques simply aren't up to the task. Clearly, we need something new. Not just a new technology, or even a new set of techniques, but an entirely new way of thinking about how we understand our systems.

What Is Observability?

Observability is the subject of an awful lot of buzz right now. It's kind of a big deal. But what is observability, actually? How is it different from (and how is it like) traditional monitoring and alerting with logs and metrics and tracing? Most importantly, how do we "do observability"?

Observability isn't just marketing hype, although it's easy to think that based on all the attention it's getting.

It's actually pretty simple. Observability is a system property, no different than resilience or manageability, that reflects how well a system's internal states can be inferred from knowledge of its external outputs. A system can be considered *observable* when it's possible to quickly and consistently ask novel questions about it with minimal prior knowledge and without having to reinstrument or build new code. An observable system lets you ask it questions that you haven't thought of yet.

Ultimately, observability is more than tooling, despite what some vendors may try to tell you (and sell you). You can't "buy observability" any more than you can "buy reliability." No tooling will make your system observable just because you're using it any more than a hammer will by itself make a bridge structurally sound. The tools can get you partway there, but it's up to you to apply them correctly.

This is much easier said than done, of course. Building observability into a complex system demands moving past searching for "known unknowns," and embracing the fact that we often can't even fully understand its state at a given snapshot in time. Understanding *all possible* failure (or non-failure) states in a complex system is pretty much impossible. The first step to achieving observability is to stop looking for specific, expected failure modes—the "known unknowns"—as if this isn't the case.

Why Do We Need Observability?

Observability is the natural evolution of traditional monitoring, driven by the new challenges introduced by cloud native architectures.

The first of these is simply the pure scale of many modern cloud native systems, which increasingly have too much *stuff* for our limited human brains with their limited human attention spans to handle. All of the data generated by multiple concurrently operating interconnected systems provides more things than we can reasonably watch, more data than we can reasonably process, and more correlations than we can reasonably make.

More importantly, however, is that the nature of cloud native systems is fundamentally different from the more traditional architectures of not-so-long-ago. Their environmental and functional requirements are different, the way they function—and the way they fail—is different, and the guarantees they need to provide are different.

How do you monitor distributed systems given the ephemerality of modern applications and the environments in which they reside? How can you pinpoint a defect in a single component within the complex web of a highly distributed system? These are the problems that "observability" seeks to address.

How Is Observability Different from "Traditional" Monitoring?

On its face, the line between monitoring and observability seems fuzzy. After all, both are about being able to ask questions of a system. The difference is in the types of questions that are and can be asked.

Traditionally, monitoring focuses on asking questions in the hope of identifying or predicting some expected or previously observed failure modes. In other words, it centers on "known unknowns." The assumption is that the system is expected to behave—and therefore fail—in a specific, predictable way. When a new failure mode is discovered—usually the hard way—its symptoms are added to the monitoring suite, and the process begins again.

This approach works well enough when a system is fairly simple, but it has some problems. First, asking a new question of a system often means writing and shipping new code. This isn't flexible, it definitely isn't scalable, and it's super annoying.

Second, at a certain level of complexity the number of "unknown unknowns" in a system starts to overwhelm the number of "known unknowns." Failures are more often unpredicted, less often predictable, and are nearly always the outcome of many things going wrong. Monitoring for every possible failure mode becomes effectively impossible.

Monitoring is something you *do to a system* to find out it isn't working. Observability techniques, on the other hand, emphasize understanding a system by allowing you to correlate events and behaviors. Observability is a *property a system has* that lets you ask why it isn't working.

The "Three Pillars of Observability"

The *Three Pillars of Observability* is the collective name by which the three most common (and foundational) tools in the observability kit—logging, metrics, and tracing—are sometimes referred. These three parts are, in the order that we'll be discussing them:

Tracing

Tracing (or *distributed tracing*) follows a request as it propagates through a (typically distributed) system, allowing the entire end-to-end request flow to be reconstructed as a directed acyclic graph (DAG) called a *trace*. Analysis of these traces can provide insight into how a system's components interact, making it possible to pinpoint failures and performance issues.

Tracing will be discussed in more detail in "Tracing" on page 350.

Metrics

Metrics involves the collection of numerical data points representing the state of various aspects of a system at specific points in time. Collections of data points, representing observations of the same subject at various times, are particularly useful for visualization and mathematical analysis, and can be used to highlight trends, identify anomalies, and predict future behavior.

We'll discuss more about metrics in "Metrics" on page 369.

Logging

Logging is the process of appending records of noteworthy events to an immutable record—the log—for later review or analysis. A log can take a variety of forms, from a continuously appended file on disk to a full-text search engine like Elasticsearch. Logs provides valuable, context-rich insight into application-specific events emitted by processes. However, it's important that log entries are properly structured; not doing so can sharply limit their utility.

We'll dive into logging in more detail in "Logging" on page 387.

While each of these methods is useful on its own, a truly observable system will interweave them so that each can reference the others. For example, metrics might be used to track down a subset of misbehaving traces, and those traces might highlight logs that could help to find the underlying cause of the behavior.

If you take nothing else away from this chapter, remember that observability is *just a system property*, like resilience or manageability, and that no tooling, framework, or vendor can "give you" observability. The so-called "Three Pillars" are just techniques that can be used to build in that property.

The (So-Called) "Three Pillars"

This name "Three Pillars of Observability" is often criticized—with good reason, I think—because it's easy to interpret it as suggesting that they're how you can "do observability." But while just having logging, metrics, and tracing won't necessarily make a system more observable, each of the "three pillars" are powerful tools that, if well understood and used well together, can provide deep insight into the internal state of your system.

Another common criticism is that the term implies that observability is the combination of three very different things, when each of the three tools simply provides a different view of the same thing, which ultimately enhances a singular ability to understand the state of your system. It's through the integration of these three approaches that it becomes possible to take the first steps towards observability.

OpenTelemetry

As of the time of writing, OpenTelemetry (or "OTel," as the cool kids are calling it[4]) is one of about four dozen "Sandbox" member projects of the Cloud Native Computing Foundation, and arguably one of the most interesting projects in the entire CNCF project catalog.

Unlike most CNCF projects, OpenTelemetry isn't a service, *per se*. Rather, it's an effort to standardize how telemetry data—traces, metrics, and (eventually) logs—are expressed, collected, and transferred. Its multiple repositories include a collection of specifications, along with APIs and reference implementations in various languages, including Go.[5]

The instrumentation space is a crowded one, with perhaps dozens of vendors and tools that have come and gone over the years, each with their own unique implementations. OpenTelemetry seeks to unify this space—and all of the vendors and tools within it—around a single vendor-neutral specification that standardizes how telemetry data is collected and sent to backend platforms. There have been other attempts to standardize before. In fact, OpenTelemetry is the merger of two such

4 I'm not one of the cool kids.

5 In addition to Go, implementations exist for Python, Java, JavaScript, .NET, C++, Rust, PHP, Erlang/Elixir, Ruby, and Swift.

earlier projects: OpenTracing and OpenCensus, which it unifies and extends into a single set of vendor-neutral standards.

In this chapter, we'll review each of the "three pillars," their core concepts, and how to use OpenTelemetry to instrument your code and forward the resulting telemetry to a backend of your choice. However, it's important to note that OpenTelemetry is a *big* subject that deserves a book of its own to truly do it justice, but I'll do my best to provide sufficient coverage to at least make it a practical introduction. At the time of this writing, there weren't any comprehensive resources available about OpenTelemetry, but I've gathered what I can from examples and a handful of articles (and a fair amount of digging through the source code).

As I was writing this chapter, I learned that Charity Majors[6] and Liz Fong-Jones were hard at work on *Observability Engineering*, planned for release by O'Reilly Media in January 2022.

The OpenTelemetry Components

OpenTelemetry extends and unifies earlier attempts at creating telemetry standards, in part by including abstractions and extension points in the SDK where you can insert your own implementations. This makes it possible to, for example, implement custom exporters that can interface with a vendor of your choice.

To accomplish this level of modularity, OpenTelemetry was designed with the following core components:

Specifications
: The OpenTelemetry specifications describe the requirements and expectations for all OpenTelemetry APIs, SDKs, and data protocols.

API
: Language-specific interfaces and implementations based on the specifications that can be used to add OpenTelemetry to an application.

SDK
: The concrete OpenTelemetry implementations that sit between the APIs and the Exporters, providing functionality like (for example) state tracking and batching data for transmission. An SDK also offers a number of configuration options for behaviors like request filtering and transaction sampling.

6 If you've never seen Charity Majors' blog, I recommend that you check it out immediately. It's one part genius plus one part experience, tied together with rainbows, cartoon unicorns, and a generous helping of rude language.

Exporters

In-process SDK plug-ins that are capable of sending data to a specific destination, which may be local (such as a log file or `stdout`), or remote (such as Jaeger, or a commercial solution like Honeycomb or Lightstep). Exporters decouple the instrumentation from the backend, making it possible to change destinations without having to reinstrument your code.

Collector

An optional, but very useful, vendor-agnostic service that can receive and process telemetry data before forwarding it to one or more destinations. It can be run either as a sidecar process alongside your application or as a standalone proxy elsewhere, providing greater flexibility for sending the application telemetry. This can be particularly useful in the kind of tightly controlled environments that are common in the enterprise.

You may have noticed the absence of an OpenTelemetry backend. Well, there isn't one. OpenTelemetry is only concerned with the collection, processing, and sending of telemetry data, and relies on you to provide a telemetry backend to receive and store the data.

There are other components as well, but the above can be considered to be OpenTelemetry's core components. The relationships between them are illustrated in Figure 11-1.

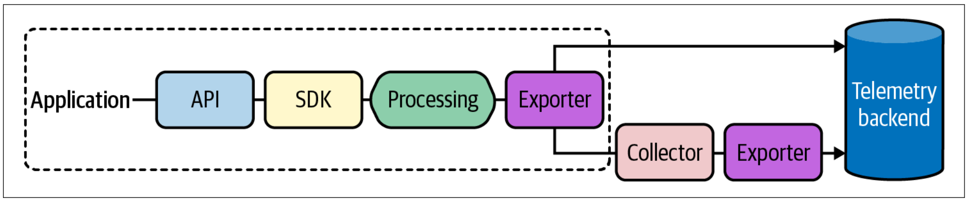


Figure 11-1. A high-level view of OpenTelemetry's core components for data instrumentation (API), processing (SDK), and exporting (exporter and collectors); you have to bring your own backend

Finally, broad language support is a central aim of the project. As of the time of this writing, OpenTelemetry provides APIs and SDKs for Go, Python, Java, Ruby, Erlang, PHP, JavaScript, .NET, Rust, C++, and Swift.

Tracing

Throughout this book, we've spent a lot of time talking about the benefits of microservices architectures and distributed systems. But the unfortunate reality—as I'm sure has already become clear—is that such architectures also introduce a variety of new and "interesting" problems.

It's been said that fixing an outage in a distributed system can feel like solving a murder mystery, which is a glib way of saying that when *something* isn't working, *somewhere* in the system, it's often a challenge just knowing where to start looking for the source of the problem before you can find and fix it.

This is exactly the kind of problem that *tracing* was invented to solve. By tracking requests as they propagate through the system—even across process, network, and security boundaries—tracing can help you to (for example) pinpoint component failures, identify performance bottlenecks, and analyze service dependencies.

Tracing is usually discussed in the context of distributed systems, but a complex monolithic application can also benefit from tracing, especially if it contends for resources like network, disk, or mutexes.

In this section, we'll go into more depth on tracing, its core concepts, and how to use OpenTelemetry to instrument your code and forward the resulting telemetry to a backend of your choice.

Unfortunately, the constraints of time and space permit us to only dig so far into this topic. But if you'd like to learn more about tracing, you might be interested in *Distributed Tracing in Practice* by Austin Parker, Daniel Spoonhower, Jonathan Mace, Ben Sigelman, and Rebecca Isaacs (O'Reilly).

Tracing Concepts

When discussing tracing, there are two fundamental concepts you need to know about, *spans* and *traces*:

Spans
: A span describes a unit of work performed by a request, such as a fork in the execution flow or hop across the network, as it propagates through a system. Each span has an associated name, a start time, and a duration. They can be (and typically are) nested and ordered to model causal relationships.

Traces
: A trace represents all of the events—individually represented as spans—that make up a request as it flows through a system. A trace may be thought of as a

directed acyclic graph (DAG) of spans, or more concretely as a "stack trace" in which each span represents the work done by one component.

This relationship between a request trace and spans is illustrated in Figure 11-2, in which we see two different representations of the same request as it flows through five different services to generate five spans.

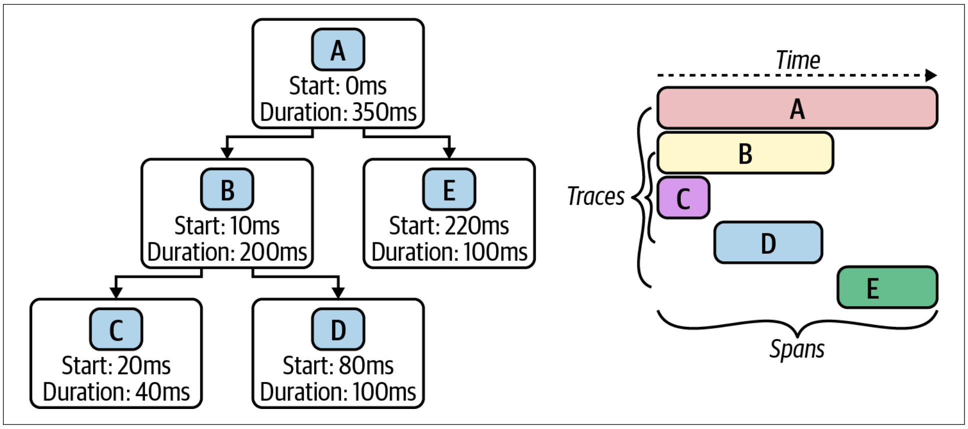


Figure 11-2. Two representations of a trace of a request as it traverses five services, resulting in five spans; the full traces are visualized as a DAG (left), and as a bar diagram (right) with a time axis illustrating start times and durations

When a request begins in the first (edge) service, it creates the first span—the *root span*—which will form the first node in the span trace. The root span is automatically assigned a globally unique trace ID, which is passed along with each subsequent hop in the request lifecycle. The next point of instrumentation creates a new span with the provided trace ID, perhaps choosing to insert or otherwise enrich the metadata associated with the request, before sending the trace ID along again with the next request.

Each hop along the flow is represented as one span. When the execution flow reaches the instrumented point at one of these services, a record is emitted with any metadata. These records are usually asynchronously logged to disk before being submitted out of band to a collector, which can then reconstruct the flow of execution based on different records emitted by different parts of the system.

Figure 11-2 demonstrates the two most common ways of illustrating a trace containing five spans, lettered A through E in the order that they were created. On the left side, the trace is represented in DAG form; the root span A starts at time 0 and lasts for 350ms, until the response is returned for the last service E. On the right, the same data is illustrated as a bar diagram with a time axis, in which the position and length of the bars reflect the start times and durations, respectively.

Tracing with OpenTelemetry

Using OpenTelemetry to instrument your code includes two phases: configuration and instrumentation. This is true whether you're instrumenting for tracing or metrics (or both), although the specifics change slightly between the two. For both tracing and metric instrumentation, the configuration phase is executed exactly once in a program, usually in the `main` function, and includes the following steps:

1. The first step is to retrieve and configure the appropriate exporters for your target backends. Tracing exporters implement the `SpanExporter` interface (which in OpenTelemetry v0.17.0 is located in the `go.opentelemetry.io/otel/sdk/export/trace` package, often aliased to `export`). As we'll discuss in "Creating the tracing exporters" on page 353, several stock exporters are included with OpenTelemetry, but custom implementations exist for many telemetry backends.
2. Before instrumenting your code for tracing, the exporters—and any other appropriate configuration options—are passed to the SDK to create the "tracer provider," which, as we'll show in "Creating a tracer provider" on page 355, will serve as the main entry point for the OpenTelemetry tracing API for the lifetime of your program.
3. Once you've created your tracer provider, it's a good practice to set it as your "global" tracer provider. As we'll see in "Setting the global tracer provider" on page 356, this makes it discoverable via the `otel.GetTracerProvider` function, which allows libraries and other dependencies that also use the OpenTelemetry API to more easily discover the SDK and emit telemetry data.

Once the configuration is complete, instrumenting your code requires only a few small steps:

1. Before you can instrument an operation, you first have to obtain a `Tracer`, which has the central role of keeping track of trace and span information, from the (usually global) tracer provider. We'll discuss this in more detail in "Obtaining a tracer" on page 356.
2. Once you have a handle to your `Tracer` you can use it to create and start the `Span` value that is the actual value that you'll use to instrument your code. We'll cover this in some detail in "Starting and ending spans" on page 357.
3. Finally, you can also choose to add metadata to your spans, including human-readable, timestamped messages called *events*, and key/value pairs called *attributes*. We'll cover span metadata in "Setting span metadata" on page 358.

OpenTelemetry Tracing Imports

There are many, many packages in the OpenTelemetry framework. Fortunately, for the purposes of this section, we'll be able to focus on just a subset of these.

The examples in this section were created using OpenTelemetry v0.17.0, which was the latest release at the time of writing. If you choose to follow along with the code presented in this section, you'll need to import the following packages from that release:

```
import (
    "go.opentelemetry.io/contrib/instrumentation/net/http/otelhttp"
    "go.opentelemetry.io/otel"
    "go.opentelemetry.io/otel/exporters/stdout"
    "go.opentelemetry.io/otel/exporters/trace/jaeger"
    "go.opentelemetry.io/otel/label"
    export "go.opentelemetry.io/otel/sdk/export/trace"
    sdktrace "go.opentelemetry.io/otel/sdk/trace"
    "go.opentelemetry.io/otel/trace"
)
```

As usual, the complete code examples are available in the GitHub repository associated with this book.

Creating the tracing exporters

The first thing you have to do when using OpenTelemetry is create and configure your exporters. Tracing exporters implement the `SpanExporter` interface, which in OpenTelemetry v0.17.0 lives in the `go.opentelemetry.io/otel/sdk/export/trace` package, which is often aliased to `export` to reduce package naming collisions.

You may recall from "The OpenTelemetry Components" on page 348 that OpenTelemetry exporters are in-process plug-ins that know how to convert metric or trace data and send it to a particular destination. This destination may be local (`stdout` or a log file) or remote (such as Jaeger, or a commercial solution like Honeycomb or Lightstep).

If you want to do anything worthwhile with the instrumentation data you collect, you'll need at least one exporter. One is usually enough, but you can define as many as you like, should you have the need. Exporters are instantiated and configured once at program startup, before being passed to the OpenTelemetry SDK. This will be covered in more detail in "Creating a tracer provider" on page 355.

OpenTelemetry comes with a number of included exporters for both tracing and metrics. Two of these are demonstrated in the following.

The Console Exporter. OpenTelemetry's Console Exporter allows you to write telemetry data as JSON to standard output. This is very handy for debugging or writing to log files. The Console Exporter is noteworthy in that it can also be used to export metric telemetry, as we'll see in "Metrics" on page 369.

Creating an instance of the Console Exporter is just a matter of calling `stdout.NewExporter`, which in OpenTelemetry v0.17.0 lives in the `go.opentelemetry.io/otel/exporters/stdout` package.

Like most exporters' creation functions, `stdout.NewExporter`, is a variadic function that can accept zero or more configuration options. We demonstrate with one of these—the option to "pretty-print" its JSON output—here:

```
stdExporter, err := stdout.NewExporter(
    stdout.WithPrettyPrint(),
)
```

In the preceding snippet, we use the `stdout.NewExporter` function, which returns both our exporter and an `error` value. We'll see what its output looks like when we run our example in "Putting It All Together: Tracing" on page 363.

For more information about the Console Exporter, please refer to its page in the relevant OpenTelemetry documentation.

The Jaeger Exporter. The Console Exporter may be useful for logging and debugging, but OpenTelemetry also includes a number of exporters designed to forward data to specialized backends, such as the Jaeger Exporter.

The Jaeger Exporter (as its name suggests) knows how to encode tracing telemetry data to the Jaeger distributed tracing system. You can retrieve an exporter value using the `jaeger.NewRawExporter` function, as shown here:

```
jaegerEndpoint := "http://localhost:14268/api/traces"
serviceName := "fibonacci"

jaegerExporter, err := jaeger.NewRawExporter(
    jaeger.WithCollectorEndpoint(jaegerEndpoint),
    jaeger.WithProcess(jaeger.Process{
        ServiceName: serviceName,
    }),
)
```

In OpenTelemetry v0.17.0, the Jaeger Exporter can be found in the `go.opentelemetry.io/otel/exporter/trace/jaeger` package.

You may have noticed that `jaeger.NewRawExporter` works a lot like `stdout.NewExporter` in that it's a variadic function that accepts zero or more configuration options, returning an `export.SpanExporter` (the Jaeger Exporter) and an `error` value.

The options passed to `jaeger.NewRawExporter` are:

- `jaeger.WithCollectorEndpoint`, which is used to define the URL that points to the target Jaeger process's HTTP collector endpoint
- `jaeger.WithProcess`, which allows you to set information about the exporting process, in this case the service's name

There are quite a few other configuration options available, but only two are used for the sake of brevity. If you're interested in more detail, please refer to its page in the relevant OpenTelemetry documentation.

What Is Jaeger?

Jaeger is an open source distributed tracing system inspired by Google's seminal 2010 paper[7] describing its Dapper distributed systems tracing infrastructure, and the earlier OpenZipkin project.

Originally developed (in Go) as an internal project by Uber Technologies, it was released as open source under the Apache license in November 2016. In September of 2017, it became the Cloud Native Computing Foundation's 12th hosted project, and advanced to graduated status in October 2019.

Among its features is support for multiple storage backends (including in-memory storage for testing setups) and a modern web UI.

Creating a tracer provider

In order to generate traces, you first have to create and initialize a *tracer provider*, represented in OpenTelemetry by the `TracerProvider` type. In OpenTelemetry v0.17.0, it lives in the `go.opentelemetry.io/otel/sdk/trace` package, which is usually aliased to `sdktrace` to avoid naming collisions.

A `TracerProvider` is a stateful value that serves as the main entry point for the OpenTelemetry tracing API, including, as we'll see in the next section, providing access to the `Tracer` that in turn serves as the provider for new `Span` values.

7 Sigelman, Benjamin H., et al. "Dapper, a Large-Scale Distributed Systems Tracing Infrastructure." *Google Technical Report*, Apr. 2010. *https://oreil.ly/Vh7Ig*.

To create a tracer provider, we use the `sdktrace.NewTracerProvider` function:

```
tp := sdktrace.NewTracerProvider(
    sdktrace.WithSyncer(stdExporter),
    sdktrace.WithSyncer(jaegerExporter))
```

In this example, the two exporters that we created in "Creating the tracing exporters" on page 353—`stdExporter` and `jaegerExporter`—are provided to `sdktrace.NewTracerProvider`, instructing the SDK to use them for exporting telemetry data.

There are several other options that can be provided to `sdktrace.NewTracerProvider`, including defining a `Batcher` or a `SpanProcessor`. These are (reluctantly) beyond the scope of this book, but more information on these can be found in the OpenTelemetry SDK Specification.

Setting the global tracer provider

Once you've created your tracer provider, it's generally a good practice to set it as your global tracer provider via the `SetTracerProvider` function. In OpenTelemetry v0.17.0, this and all of OpenTelemetry's global options live in the `go.opentelemetry.io/otel` package.

Here we set the global tracer provider to be the value of `tp`, which we created in the previous section:

```
otel.SetTracerProvider(tp)
```

Setting the global tracer provider makes it discoverable via the `otel.GetTracerProvider` function. This allows libraries and other dependencies that use the OpenTelemetry API to more easily discover the SDK and emit telemetry data:

```
gtp := otel.GetTracerProvider(tp)
```

If you don't explicitly set a global tracer provider, `otel.GetTracerProvider` will return a no-op `TracerProvider` implementation that returns a no-op `Tracer` that provides no-op `Span` values.

Obtaining a tracer

In OpenTelemetry, a `Tracer` is a specialized type that keeps track of trace and span information, including what span is currently active. Before you can instrument an operation you first have to use a (usually global) tracer provider's `Tracer` method to obtain a `trace.Tracer` value:

```
tr := otel.GetTracerProvider().Tracer("fibonacci")
```

TracerProvider's `Tracer` method accepts a string parameter to set its name. By convention, Tracers are named after the component they are instrumenting, usually a library or a package.

Now that you have your tracer, your next step will be to use it to create and start a new `Span` instance.

Starting and ending spans

Once you have a handle to a `Tracer`, you can use it to create and start new `Span` values representing named and timed operations within a traced workflow. In other words, a `Span` value represents the equivalent of one step in a stack trace.

In OpenTelemetry v0.17.0, both the `Span` and `Tracer` interfaces can be found in the `go.opentelemetry.io/otel/trace`. Their relationship can be deduced by a quick review of Tracer's definition code:

```
type Tracer interface {
    Start(ctx context.Context, spanName string, opts ...trace.SpanOption)
        (context.Context, trace.Span)
}
```

Yes, that's really all there is. Tracer's only method, `Start`, accepts three parameters: a `context.Context` value, which is the mechanism that `Tracer` uses to keep track of spans; the name of the new span, which by convention is usually the name of the function or component being evaluated; and zero or more span configuration options.

Unfortunately, a discussion of the available span configurations is beyond the scope of this book, but if you're interested, more detail is available in the relevant Go Documentation.

Importantly, `Start` returns not just the new `Span`, but also a `context.Context`. This is a new `Context` instance derived from the one that was passed in. As we'll see shortly, this is important when we want to create child `Span` values.

Now that you have all of the pieces in place, you can begin instrumenting our code. To do this, you request a `Span` value from your `Tracer` via its `Start` method, as shown in the following:

```
const serviceName = "foo"

func main() {
    // EXPORTER SETUP OMITTED FOR BREVITY

    // Retrieve the Tracer from the otel TracerProvider.
    tr := otel.GetTracerProvider().Tracer(serviceName)

    // Start the root span; receive a child context (which now
    // contains the trace ID), and a trace.Span.
    ctx, sp := tr.Start(context.Background(), "main")
    defer sp.End()     // End completes the span.

    SomeFunction(ctx)
}
```

In this snippet we use Tracer's `Start` method to create and start a new `Span`, which returns a derived context and our `Span` value. It's important to note that we ensure that the `Span` is ended by calling it in a `defer`, so that `SomeFunction` is entirely captured in the root `Span`.

Of course, we'll also want to instrument `SomeFunction`. Since it receives the derived context we got from the original `Start`, it can now use that `Context` to create its own subspan:

```
func SomeFunction(ctx context.Context) {
    tr := otel.GetTracerProvider().Tracer(serviceName)
    _, sp := tr.Start(ctx, "SomeFunction")
    defer sp.End()

    // Do something MAGICAL here!
}
```

The only differences between `main` and `SomeFunction` are the names of the spans and the `Context` values. It's significant that `SomeFunction` uses the `Context` value derived from the original `Start` call in `main`.

Setting span metadata

Now that you have a `Span`, what do you do with it?

If you do nothing at all, that's okay. As long as you've remembered to `End` your `Span` (preferably in a `defer` statement) a minimal timeline for your function will be collected.

However, the value of your span can be enhanced with the addition of two types of metadata: *attributes* and *events*.

Attributes. Attributes are key/value pairs that are associated with spans. They can be used later for aggregating, filtering, and grouping traces.

If known ahead of time, attributes can be added when a span is created by passing them as option parameters to the `tr.Start` method using the `WithAttributes` function:

```
ctx, sp := tr.Start(ctx, "attributesAtCreation",
    trace.WithAttributes(
        label.String("hello", "world"), label.String("foo", "bar")))
defer sp.End()
```

Here we call `tr.Start` to start a new span, passing it our active `context.Context` value and a name. But `Start` is also a variadic function that can accept zero or more options, so we opt to use the `WithAttributes` function to pass two string attributes: `hello=world` and `foo=far`.

The `WithAttributes` function accepts a `label.KeyValue` type, from OpenTelemetry's `go.opentelemetry.io/otel/label` package. Values of this type can be created using the various type methods, such as `label.String` as above. Methods exist for all Go types (and more). See the label package's documentation for more information.

Attributes don't have to be added at span creation time. They can be added later in a span's lifecycle as well, as long as the span hasn't yet been completed:

```
answer := LifeTheUniverseAndEverything()
span.SetAttributes(label.Int("answer", answer))
```

Events. An *event* is a timestamped, human-readable message on a span that represents *something* happening during the span's lifetime.

For example, if your function requires exclusive access to a resource that's under a mutex, you might find it useful to add events when you acquire and release the lock:

```
span.AddEvent("Acquiring mutex lock")
mutex.Lock()

// Do something amazing.

span.AddEvent("Releasing mutex lock")
mutex.Unlock()
```

If you like, you can even add attributes to your events:

```
span.AddEvent("Canceled by external signal",
    label.Int("pid", 1234),
    label.String("signal", "SIGHUP"))
```

Autoinstrumentation

Autoinstrumentation, broadly, refers to instrumentation code that you didn't write. This is a useful feature that can spare you from a fair amount of unnecessary bookkeeping.

OpenTelemetry supports autoinstrumentation through various wrappers and helper functions around many popular frameworks and libraries, including ones that we cover in this book, like `net/http`, `gorilla/mux`, and `grpc`.

While using these functionalities doesn't free you from having to configure OpenTelemetry at startup, they do remove some of the effort associated with having to manage your traces.

Autoinstrumenting net/http and gorilla/mux. In OpenTelemetry 0.17.0, autoinstrumentation support for both the standard `net/http` library and `gorilla/mux`, both of which we first covered in Chapter 5 in the context of building a RESTful web service, is provided by the `go.opentelemetry.io/contrib/instrumentation/net/http/otelhttp` package.

Its use is refreshingly minimalist. Take, for example, this standard idiom in `net/http` for registering a handler function to the default mux[8] and starting the HTTP server:

```
func main() {
    http.HandleFunc("/", helloGoHandler)
    log.Fatal(http.ListenAndServe(":3000", nil))
}
```

In OpenTelemetry, a handler function can be autoinstrumented by passing it to the `otelhttp.NewHandler` function, the signature for which is shown here:

```
func NewHandler(handler http.Handler, operation string, opts ...Option)
    http.Handler
```

The `otelhttp.NewHandler` function accepts and returns a handler function. It works by wrapping the passed handler function in a second handler function that creates a span with the provided name and options, so that the original handler acts like middleware within the returned span-handling function.

A typical application of the `otelhttp.NewHandler` function is shown in the following:

```
func main() {
    http.Handle("/",
        otelhttp.NewHandler(http.HandlerFunc(helloGoHandler), "root"))
    log.Fatal(http.ListenAndServe(":3000", nil))
}
```

You'll notice that we have to cast the handler function to a `http.HandlerFunc` before passing it to `otelhttp.NewHandler`. This wasn't necessary before because `http.HandleFunc` performs this operation automatically before itself calling `http.Handle`.

If you're using `gorilla/mux`, the change is almost the same, except that you're using the `gorilla` mux instead of the default mux:

```
func main() {
    r := mux.NewRouter()
    r.Handle("/",
```

8 Recall that the name "mux" is short for "HTTP request multiplexer."

```
        otelhttp.NewHandler(http.HandlerFunc(helloGoHandler), "root"))
    log.Fatal(http.ListenAndServe(":3000", r))
}
```

You'll need to repeat this for each handler function you want to instrument, but either way the total amount of code necessary to instrument your entire service is pretty minimal.

Autoinstrumenting gRPC. In OpenTelemetry 0.17.0, autoinstrumentation support for gRPC, which we introduced in Chapter 8 in the context of loosely coupled data interchange, is provided by the `go.opentelemetry.io/contrib/instrumentation/google.golang.org/grpc/otelgrpc` package.[9]

Just like autoinstrumentation for `net/http`, autoinstrumentation for gRPC is very minimalist, leveraging *gRPC interceptors*. We haven't talked about gRPC interceptors at all yet, and unfortunately a full treatment of gRPC interceptors is beyond the scope of this book. They can be described as the gRPC equivalent to middleware in `gorilla/mux`, which we leveraged in "Load shedding" on page 274 to implement automatic load shedding.

As their name implies, gRPC interceptors can intercept gRPC requests and responses to, for example, inject information into the request, update the response before it's returned to the client, or to implement a cross-cutting functionality like authorization, logging, or caching.

If you'd like to learn a little more about gRPC interceptors, the article "Interceptors in gRPC-Web" on the gRPC blog offers a good introduction to the subject. For a more in-depth coverage, you might want to invest in a copy of *gRPC: Up and Running* by Kasun Indrasiri and Danesh Kuruppu (O'Reilly).

Taking a look at a slice of the original service code from "Implementing the gRPC service" on page 236, you can see two of the operative functions:

```
s := grpc.NewServer()
pb.RegisterKeyValueServer(s, &server{})
```

In the above snippet, we create a new gRPC server, and pass that along to our autogenerated code package to register it.

Interceptors can be added to a gRPC server using the `grpc.UnaryInterceptor` and/or `grpc.StreamInterceptor`, the former of which is used to intercept unary

9 That wins the record for longest package name, at least in this book.

(standard request–response) service methods, and the latter of which is used for intercepting streaming methods.

To autoinstrument your gRPC server, you use one or both of these functions to add one or more off-the-shelf OpenTelemetry interceptors, depending on the types of requests your service handles:

```
s := grpc.NewServer(
    grpc.UnaryInterceptor(otelgrpc.UnaryServerInterceptor()),
    grpc.StreamInterceptor(otelgrpc.StreamServerInterceptor()),
)

pb.RegisterKeyValueServer(s, &server{})
```

While the service we built in Chapter 8 uses exclusively unary methods, the preceding snippet adds interceptors for both unary and stream methods for the sake of demonstration.

Getting the current span from context. If you're taking advantage of autoinstrumentation, a trace will automatically be created for each request. While convenient, this also means that you don't have your current `Span` immediately on hand for you to enhance with application-specific attribute and event metadata. So, what do you do?

Fear not! Since your application framework has conveniently placed the span data inside the current context, the data is easily retrievable:

```
func printSpanHandler(w http.ResponseWriter, req *http.Request) {
    ctx := req.Context()                    // Get the request Context

    span := trace.SpanFromContext(ctx)      // Get the current span

    fmt.Printf("current span: %v\n", span)  // Why not print the span?
}
```

What About the Clients?

For a complete distributed tracing experience, tracing metadata has to be forwarded across service boundaries. That means you have to instrument on the client side as well.

The good news is that OpenTelemetry supports autoinstrumentation of `http` and gRPC clients.

Unfortunately, there isn't enough space available in this chapter to cover client autoinstrumentation in any detail. Examples of both `http` and `gRPC` client autoinstrumentation are available in the GitHub repository associated with this book, however.

Putting It All Together: Tracing

Using all the parts that we've discussed in this section, let's now build a small web service. Because we're going to instrument this service with tracing, the ideal service would make a whole lot of function calls, but would still be pretty small.

We're going to build a Fibonacci service. Its requirements are very minimal: it will be able to accept an HTTP GET request, in which the *n*th Fibonacci number can be requested using parameter `n` on the GET query string. For example, to request the sixth Fibonacci number, you should be able to `curl` the service as: `http://localhost:3000?n=6`.

To do this, we'll use a total of three functions. Starting from the inside and working our way out, these are:

The service API
: This will do the Fibonacci computation proper—at the request of the service handler—by recursively calling itself, with each call generating its own span.

The service handler
: This is an HTTP handler function as defined by the `net/http` package, which will be used just like in "Building an HTTP Server with net/http" on page 112 to receive the client request, call the service API, and return the result in the response.

The main function
: In the `main` function, the OpenTelemetry exporters are created and registered, the service handler function is provided to the HTTP framework, and the HTTP server is started.

The Fibonacci service API

The service API at the very core of the service is where the actual computation is performed. In this case, it's a concurrent implementation of the Fibonacci method to calculate the *n*th Fibonacci number.

Just like any good service API, this function doesn't know (or care) how it's being used, so it has no knowledge of HTTP requests or responses:

```go
func Fibonacci(ctx context.Context, n int) chan int {
    ch := make(chan int)

    go func() {
        tr := otel.GetTracerProvider().Tracer(serviceName)

        cctx, sp := tr.Start(ctx,
            fmt.Sprintf("Fibonacci(%d)", n),
            trace.WithAttributes(label.Int("n", n)))
        defer sp.End()

        result := 1
        if n > 1 {
            a := Fibonacci(cctx, n-1)
            b := Fibonacci(cctx, n-2)
            result = <-a + <-b
        }

        sp.SetAttributes(label.Int("result", result))

        ch <- result
    }()

    return ch
}
```

In this example, the `Fibonacci` function doesn't know how it's being used, but it *does* know about the OpenTelemetry package. Autoinstrumentation can only trace what it wraps. Anything within the API will need to instrument itself.

This function's use of `otel.GetTracerProvider` ensures that it'll get the global `TracerProvider`, assuming that it was configured by the consumer. If no global tracer provider has been set, these calls will be no-ops.

For extra credit, take a minute to add support for `Context` cancellation to the `Fibonacci` function.

The Fibonacci service handler

This is an HTTP handler function as defined by the `net/http` package.

It'll be used in our service just like in "Building an HTTP Server with net/http" on page 112: to receive the client request, call the service API, and return the result in the response:

```
func fibHandler(w http.ResponseWriter, req *http.Request) {
    var err error
    var n int

    if len(req.URL.Query()["n"]) != 1 {
        err = fmt.Errorf("wrong number of arguments")
    } else {
        n, err = strconv.Atoi(req.URL.Query()["n"][0])
    }

    if err != nil {
        http.Error(w, "couldn't parse index n", 400)
        return
    }

    // Retrieve the current context from the incoming request
    ctx := req.Context()

    // Call the child function, passing it the request context.
    result := <-Fibonacci(ctx, n)

    // Get the Span associated with the current context and
    // attach the parameter and result as attributes.
    if sp := trace.SpanFromContext(ctx); sp != nil {
        sp.SetAttributes(
            label.Int("parameter", n),
            label.Int("result", result))
    }

    // Finally, send the result back in the response.
    fmt.Fprintln(w, result)
}
```

Note that it doesn't have to create or end a `Span`; autoinstrumentation will do that for us.

It *does*, however, set some attributes on the current span. To do this, it uses `trace.SpanFromContext` to retrieve the current span from the request context. Once it has the span, it's free to add whatever metadata it likes.

The `trace.SpanFromContext` function will return `nil` if it can't find a `Span` associated with the `Context` passed to it.

The service Main function

At this point, all of the hard work has been done. All we have left to do is configure OpenTelemetry, register the handler function with the default HTTP mux, and start the service:

```
const (
    jaegerEndpoint = "http://localhost:14268/api/traces"
    serviceName    = "fibonacci"
)

func main() {
    // Create and configure the console exporter
    stdExporter, err := stdout.NewExporter(
        stdout.WithPrettyPrint(),
    )
    if err != nil {
        log.Fatal(err)
    }

    // Create and configure the Jaeger exporter
    jaegerExporter, err := jaeger.NewRawExporter(
        jaeger.WithCollectorEndpoint(jaegerEndpoint),
        jaeger.WithProcess(jaeger.Process{
            ServiceName: serviceName,
        }),
    )
    if err != nil {
        log.Fatal(err)
    }

    // Create and configure the TracerProvider exporter using the
    // newly created exporters.
    tp := sdktrace.NewTracerProvider(
        sdktrace.WithSyncer(stdExporter),
        sdktrace.WithSyncer(jaegerExporter))

    // Now we can register tp as the otel trace provider.
    otel.SetTracerProvider(tp)

    // Register the autoinstrumented service handler
    http.Handle("/",
        otelhttp.NewHandler(http.HandlerFunc(fibHandler), "root"))

    // Start the service listening on port 3000
    log.Fatal(http.ListenAndServe(":3000", nil))
}
```

As you can see, the majority of the main method is dedicated to creating our (console and Jaeger) exporters and configuring the tracer provider as we did in "Creating the tracing exporters" on page 353. Note the value of `jaegerEndpoint`, which assumes that you'll have a local Jaeger service running. We'll do that in the next step.

The last two lines are spent autoinstrumenting and registering the handler function and starting the HTTP service, just as we did in "Autoinstrumentation" on page 359.

Starting your services

Before we continue, we'll want to start a Jaeger service to receive the telemetry data provided by the Jaeger exporter that we included. For a little more background on Jaeger, see "What Is Jaeger?" on page 355.

If you have Docker installed, you can start a Jaeger service with the following command:

```
$ docker run -d --name jaeger   \
  -p 16686:16686                \
  -p 14268:14268                 \
  jaegertracing/all-in-one:1.21
```

Once the service is up and running, you'll be able to access its web interface by browsing to `http://localhost:16686`. Obviously, there won't be any data there yet, though.

Now for the fun part: start your service by running its main function:

```
$ go run .
```

Your terminal should pause. As usual, you can stop the service with a Ctrl-C.

Finally, in another terminal, you can now send a request to the service:

```
$ curl localhost:3000?n=6
13
```

After a short pause, you should be rewarded with a result. In this case, 13.

Be careful with the value of `n`. If you make it `n` too large, it might take the service a long time to respond, or even crash.

Console exporter output

Now that you've issued a request to your service, take a look at the terminal you used to start your service. You should see several JSON blocks that resemble the following:

```
[
    {
        "SpanContext":{
            "TraceID":"4253c86eb68783546b8ae3b5e59b4a0c",
            "SpanID":"817822981fc2fb30",
            "TraceFlags":1
        },
        "ParentSpanID":"0000000000000000",
        "SpanKind":1,
        "Name":"main",
        "StartTime":"2020-11-27T13:50:29.739725-05:00",
        "EndTime":"2020-11-27T13:50:29.74044542-05:00",
        "Attributes":[
            {
                "Key":"n",
                "Value":{
                    "Type":"INT64",
                    "Value":6
                }
            },
            {
                "Key":"result",
                "Value":{
                    "Type":"INT64",
                    "Value":13
                }
            }
        ],
        "ChildSpanCount":1,
        "InstrumentationLibrary":{
            "Name":"fibonacci",
            "Version":""
        }
    }
]
```

These JSON objects are the output of the Console Exporter (which, remember, we've configured to pretty-print). There should be one per span, which is quite a few.

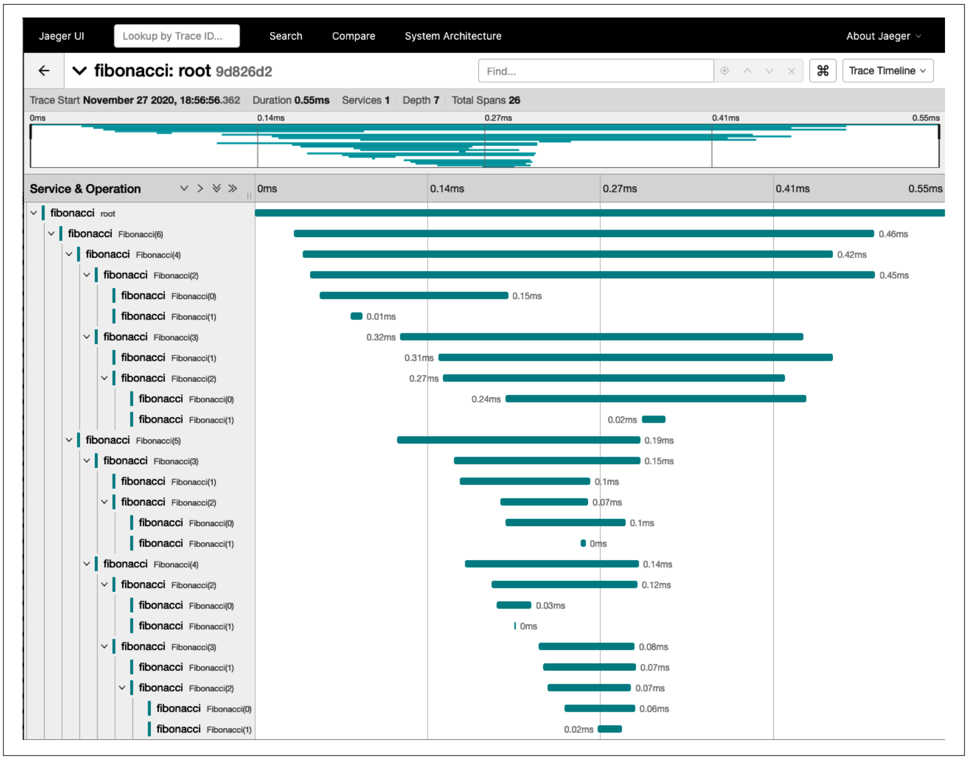
The preceding example (which has been pruned slightly) is from the root span. As you can see, it includes quite a few interesting bits of data, including its start and end times, and its trace and span IDs. It even includes the two attributes that we explicitly set: the input value `n`, and the result of our query.

Viewing your results in Jaeger

Now that you've generated your trace and sent it to Jaeger, it's time to visualize it. Jaeger just happens to provide a slick web UI for exactly that purpose!

To check it out, browse to `http://localhost:16686` with your favorite web browser. Select Fibonacci in the Service dropdown, and click the Find traces button. You should be presented with output similar to that shown in Figure 11-3.

Each bar in the visualization represents a single span. You can even view a specific span's data by clicking on it, which reveals the same data that was contained in the (quite verbose) console output that you saw in "Console exporter output" on page 367.



Jaeger UI
Lookup by Trace ID...
Search
Compare
System Architecture
About Jaeger
fibonacci: root 9d826d2
Find...
Trace Timeline
Trace Start November 27 2020, 18:56:56.362 Duration 0.55ms Services 1 Depth 7 Total Spans 26
0ms
0.14ms
0.27ms
0.41ms
0.55ms
Service & Operation
fibonacci root
fibonacci Fibonacci(6) 0.46ms
fibonacci Fibonacci(4) 0.42ms
fibonacci Fibonacci(2) 0.45ms
fibonacci Fibonacci(0) 0.15ms
fibonacci Fibonacci(1) 0.01ms
fibonacci Fibonacci(3) 0.32ms
fibonacci Fibonacci(1) 0.31ms
fibonacci Fibonacci(2) 0.27ms
fibonacci Fibonacci(0) 0.24ms
fibonacci Fibonacci(1) 0.02ms
fibonacci Fibonacci(5) 0.19ms
fibonacci Fibonacci(3) 0.15ms
fibonacci Fibonacci(1) 0.1ms
fibonacci Fibonacci(2) 0.07ms
fibonacci Fibonacci(0) 0.1ms
fibonacci Fibonacci(1) 0ms
fibonacci Fibonacci(4) 0.14ms
fibonacci Fibonacci(2) 0.12ms
fibonacci Fibonacci(0) 0.03ms
fibonacci Fibonacci(1) 0ms
fibonacci Fibonacci(3) 0.08ms
fibonacci Fibonacci(1) 0.07ms
fibonacci Fibonacci(2) 0.07ms
fibonacci Fibonacci(0) 0.06ms
fibonacci Fibonacci(1) 0.02ms

Figure 11-3. Screenshot of the Jaeger interface, displaying the results of a concurrent Fibonacci call

Metrics

Metrics is the collection of numerical data about a component, process, or activity over time. The number of potential metric sources is vast, and includes (but isn't limited to) things like computing resources (CPU, memory used, disk and network I/O), infrastructure (instance replica count, autoscaling events), applications (request count, error count), and business metrics (revenue, customer sign-ups, bounce rate, cart abandonment). Of course, these are just a handful of trivial examples. For a complex system, the *cardinality* can range into the many thousands, or even millions.

A metric data point, representing one observation of a particular aspect of the target (such as the number of hits an endpoint has received), is called a *sample*. Each sample has a name, a value, and a millisecond-precision timestamp. Also—at least in modern systems like Prometheus—a set of key-value pairs called *labels*.

Cardinality

Cardinality is an important concept in observability. It has its origins in set theory, where it's defined as the number of elements in a set. For example, the set $A = \{1, 2, 3, 5, 8\}$ contains five elements, so A therefore has a cardinality of five.

The term was later adopted by database designers to refer to the number of distinct values in a table column.[10] For example, "eye color" would have a low cardinality, while "username" would be quite high.

More recently, however, the term "cardinality" has been adopted for monitoring, where it's come to refer to the number of unique combinations of metric names and dimensions—the number of distinct label values attached to a particular metric—in your monitoring system. High cardinality information is critical for observability because it means that the data has many different ways that it can be queried, making it all the more likely that you'll be able to ask it a question that you'd never thought to ask before.

By itself, a single sample is of limited use, but a sequence of successive samples with the same name and labels—a *time series*—can be incredibly useful. As illustrated in Figure 11-4, collecting samples as a time series allows metrics to be easily visualized by plotting the data points on a graph, in turn making it easier to see trends or to observe anomalies or outliers.

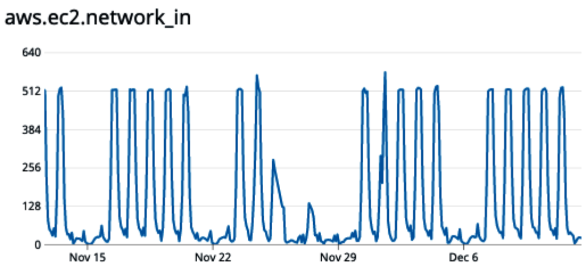


Figure 11-4. Arranging samples as a time series allows them to be graphically visualized

In the above figure, we show a time series of the metric `aws.ec2.network_in` for one AWS EC2 instance. Time is on the x-axis (specifically, one month spanning November–December 2020). The y-axis represents the instantaneous rate at which the instance is receiving network data at that moment. Visualizing the time series this way, it becomes obvious that traffic to the instance spikes each weekday.

10 It can *also* refer to the numerical relationship between two database tables (i.e., *one-to-one*, *one-to-many*, or *many-to-many*), but that definition is arguably less relevant here.

Interestingly, November 25–27—the days spanning the day before to the day after Thanksgiving in the United States—are the exceptions.

The true power of metrics, however, isn't its ability to be visually represented for human eyes: it's that its numerical nature makes it particularly amenable to mathematical modeling. For example, you might use trend analysis to detect anomalies or predict future states, which in turn can inform decisions or trigger alerts.

Push Versus Pull Metric Collection

There are two primary architectures in the universe of metrics: push-based and pull-based (so called because of the relationship between the components being monitored and the collector backend).

In push-based metrics, monitored components "push" their data to a central collector backend. In pull-based metrics, the inverse is true: the collector actively retrieves metrics by "pulling" them from HTTP endpoints exposed by the monitored components (or by sidecar services deployed for this purpose, also confusingly called "exporters"; see "Prometheus Exporters" on page 377). Both approaches are illustrated in Figure 11-5.

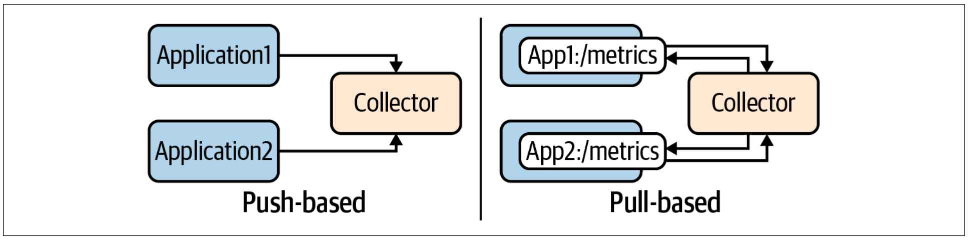


Figure 11-5. Push-based metrics (left) send telemetry directly to a central collector backend; pull-based metrics (right) are actively scraped by the collector from exposed metric endpoints

What follows is a short description of each of these approaches, along with a very limited list of some arguments for and against each approach. Unfortunately, there are bounteous arguments, many quite nuanced—far too nuanced to delve into here—so we'll have to be content with some of the common ones.

Push-based metric collection

In push-based metric collection, an application, either directly or via a parallel agent process, periodically sends data to a central collector backend. Push implementations, like Ganglia, Graphite, and StatsD, tend to be the most common (even default) approach, perhaps in part because the push model tends to be quite a bit easier to reason about.

Push messages are typically unidirectional, being emitted by the monitored components or monitoring agent and sent to a central collector. This places a bit less burden on the network relative to the (bidirectional) pull model, and can reduce the complexity of the network security model, since components don't have to make a metrics endpoint accessible to the collector. It's also easier to use the push model to monitor highly ephemeral components such as short-lived containers or serverless functions.

There are some downsides to the push model, though. First, you need to know where to send your request. While there are lots of ways of doing this, each has its downside, ranging from hardcoded addresses (which are hard to change) to DNS lookups or service discovery (which may add unacceptable latency). Scaling can also sometimes be an issue, in that it's entirely possible for a large number of components to effectively DDoS your collector backend.

Pull-based metric collection

In the pull-based collection model, the collector backend periodically (on some configurable cadence) scrapes a metric endpoint exposed by a component, or by a proxy deployed for this purpose. Perhaps the best-known example of a pull-based system is Prometheus.

The pull approach offers some notable advantages. Exposing a metric endpoint decouples the components being observed from the collector itself, which provides all of the benefits of loose coupling. For example, it becomes easier to monitor a service during development, or even manually inspect a component's health with a web browser. It's also much easier for a pull model to tell if a target is down.

However, the pull approach has a discovery issue of its own, in that the collector has to somehow know where to find the services it's supposed to monitor. This can be a bit of a challenge, particularly if your system isn't using dynamic service discovery. Load balancers are of little help here, either, since each request will be forwarded to a random instance, greatly reducing the effective collection rate (since each of N instance receives 1/N of the pulls) and severely muddying what data is collected (since all of the instances tend to look like a single target). Finally, pull-based collection can make it somewhat harder to monitor very short-lived ephemeral things like serverless functions, necessitating a solution like the push gateway.

What Is Prometheus?

Prometheus is an open source monitoring and alerting toolkit. It uses a pull model over HTTP to scrape metric data, storing it as high-dimensionality values in its time series database.

Prometheus consists of a core server, which is responsible for the acquisition and storage of data, as well as a variety of other optional components, including a push gateway for supporting data pushes by short-lived jobs, and an alertmanager for handling alerts. While Prometheus isn't intended as a dashboarding solution, it also provides a basic web UI and query language, PromQL, to make data more easily accessible.

In January of 2015, SoundCloud publicly released Prometheus as an open source project under the Apache license, and in May 2016, Prometheus joined the Cloud Native Computing Foundation as its second hosted project (after Kubernetes). It advanced to graduated status in August 2018.

But which is better?

Since the push and pull approaches are, it would seem, polar opposites of one another, it's common for people to wonder which is better.[11] That's a hard question, and as is often the case when comparing technical methodologies, the answer is a resounding "it depends."

Of course, that's never stopped a sufficiently motivated programmer from stridently arguing one side or another, but at the end of the day, the "better" approach is the one that satisfies the requirements of your system. Of course (and quite unsatisfyingly) that could be both. We technical types abhor ambiguity, yet it stubbornly insists on existing anyway.

So, I will close this section with the words of Brian Brazil, a core developer of Prometheus:

> From an engineering standpoint, in reality, the question of push versus pull largely doesn't matter. In either case, there's advantages and disadvantages, and with engineering effort, you can work around both cases.[12]

11 Whatever "better" means.

12 Kiran, Oliver. "Exploring Prometheus Use Cases with Brian Brazil." *The New Stack Makers*, 30 Oct. 2016. *https://oreil.ly/YDIek*.

Metrics with OpenTelemetry

As of the time of writing, the OpenTelemetry metrics API is still in alpha, so it still has a few rough spots to be ironed out and a few inconsistencies with the tracing API that are yet to be resolved.

That being said, the considerable private and community support behind OpenTelemetry, coupled with its quite impressive rate of development, make it appropriate not just for inclusion in this book, but as the most likely candidate to become the gold standard for metric telemetry for the next several years at least.

For the most part, OpenTelemetry metrics work a lot like traces, but are different enough to possibly cause some confusion. For both tracing and metric instrumentation, the configuration phase is executed exactly once in a program, usually in the `main` function, and includes the following steps:

1. The first step is to create and configure the appropriate exporter for the target backend. Metric exporters implement the `metric.Exporter` interface, which in OpenTelemetry v0.17.0 is located in the `go.opentelemetry.io/otel/sdk/export/metric` package. As we'll discuss in "Creating the metric exporters" on page 375, several stock exporters are included with OpenTelemetry, but unlike trace exporters, you can currently only use one metric exporter at a time.
2. Before instrumenting your code for metrics, the exporter is used to define the global "meter provider," which will serve as your program's main entry point into the OpenTelemetry metric API throughout its lifetime. As we'll see in "Setting the global meter provider" on page 376, this makes the meter exporter discoverable via the `otel.GetMeterProvider` function, which allows libraries and other dependencies that use the OpenTelemetry API to more easily access the SDK and emit telemetry data.
3. If your metric backend uses a pull-based design like Prometheus, you'll have to expose a metric endpoint that it can pull from. You'll see how the Prometheus exporter leverages Go's standard `http` package to do this in "Exposing the metrics endpoint" on page 376.

Once the configuration is complete, instrumenting your code requires only a few small steps:

1. Before you can instrument an operation, you first have to obtain a `Meter`, the structure through which all metric collection is configured and reported, from the meter provider. We'll discuss this in more detail in "Obtaining a meter" on page 378.
2. Finally, once you have a `Meter`, you can use it to instrument your code. There are two ways this can be done, either by explicitly recording measurements, or by

creating *observers* that can autonomously and asynchronously collect data. Both of these approaches are covered in "Metric instruments" on page 379.

OpenTelemetry Metrics Imports

There are many, many packages in the OpenTelemetry framework. Fortunately, for the purposes of this section, we'll be able to focus on just a subset of these.

The examples in this section were created using OpenTelemetry v0.17.0, which was the latest release at the time of writing. If you choose to follow along with this section, you'll need to import the following packages from that release:

```
import (
    "go.opentelemetry.io/otel"
    "go.opentelemetry.io/otel/exporters/metric/prometheus"
    "go.opentelemetry.io/otel/label"
    "go.opentelemetry.io/otel/metric"
)
```

As usual, the complete code examples are available in the GitHub repository associated with this book.

Creating the metric exporters

Just like with tracing, the first thing you have to do when using OpenTelemetry for metrics is create and configure your exporters. Metric exporters implement the `metric.Exporter` interface, which in OpenTelemetry v0.17.0 lives in the `go.opentelemetry.io/otel/sdk/export/metric` package.

The way that you create metric exporters varies a little between implementations, but it's typical for an exporter to have a `NewExportPipeline` builder function, at least in the standard OpenTelemetry packages.

To get an instance of the Prometheus exporter, for example, you would use the `NewExportPipeline` function from the `go.opentelemetry.io/otel/exporters/metric/prometheus` package:

```
prometheusExporter, err := prometheus.NewExportPipeline(prometheus.Config{})
```

The above snippet creates the exporter and configures it according the directions specified by the passed `prometheus.Config` value. Any behaviors not overridden by the `Config` will use the recommended options.

The `prometheus.Config` parameter also allows you to specify a variety of custom behaviors. Unfortunately, the specifics are beyond the scope of this book, but if you're interested the exporter Config code and the code for the Prometheus Go client are fairly straightforward.

Setting the global meter provider

Where OpenTelemetry tracing has the "tracer provider" that provides `Tracer` values, OpenTelemetry metrics has the *meter provider*, which provides the `Meter` values through which all metric collection is configured and reported.

You may recall that when working with tracing exporters, defining the global tracer provider requires two steps: creating and configuring a tracer provider instance, and then setting that instance as the global tracer provider.

The meter provider works a little differently: rather than using one or more exporters to create and define a provider (as is the case with the `TracerProvider`), a meter provider is typically retrieved *from* the metric exporter, and then passed directly to the `otel.SetMeterProvider` function:

```
// Get the meter provider from the exporter.
mp := prometheusExporter.MeterProvider()

// Set it as the global meter provider.
otel.SetMeterProvider(mp)
```

An unfortunate consequence of this design is that you're limited to using only one metric exporter at a time, since the meter provider is provided by the exporter instead of the other way around. Obviously, this is a significant deviation from how the tracing API works, and I expect it to change as the OpenTracing metrics API moves into beta.

There's also a `prometheus.InstallNewPipeline` convenience function that can be used instead of explicitly calling the `prometheus.NewExportPipeline` and `otel.SetMeterProvider` functions.

Exposing the metrics endpoint

Because Prometheus is pull-based, any telemetry data we want to send it must be exposed through an HTTP endpoint that the collector can scrape.

To do this, we can make use of Go's standard `http` package, which, as we've shown several times in this book, requires minimal configuration, and is rather straightforward to use.

To review what we first introduced in "Building an HTTP Server with net/http" on page 112, starting a minimal HTTP server in Go requires at least two calls:

- `http.Handle` to register a handler function that implements the `http.Handler` interface
- `http.ListenAndServe` to start the server listening

But the OpenTelemetry Prometheus exporter has a pretty nifty trick up its sleeve: it implements the `http.Handler` interface, which allows it to be passed directly to `http.Handle` to act as a handler function for the metric endpoint! See the following:

```
// Register the exporter as the handler for the "/metrics" pattern.
http.Handle("/metrics", prometheusExporter)

// Start the HTTP server listening on port 3000.
log.Fatal(http.ListenAndServe(":3000", nil))
```

In this example, we pass the Prometheus exporter directly into `http.Handle` to register it as the handler for the pattern "/metrics." It's hard to get more convenient than that.

Ultimately, the name of your metrics endpoint is up to you, but `metrics` is the most common choice. It's also where Prometheus looks by default.

Prometheus Exporters

It's relatively straightforward to expose a metrics endpoint if your application is a standard web service written in Go. But what if you want to collect JMX data from a JVM-based application, query metrics from a PostgreSQL database, or system metrics from a deployed Linux or Windows instance?

Unfortunately, not all of the things you'll want to collect data on are things that you control, and very few of them natively expose their own metrics endpoints.

In this (very common) scenario, the typical pattern is to deploy a *Prometheus exporter*. A Prometheus exporter (not to be confused with an OpenTelemetry exporter) is a specialized adapter that runs as a service to collect the desired metric data and expose it on a metrics endpoint.

As of the time of writing, there are over 200 different Prometheus exporters listed in the Prometheus documentation, and many more community-built exporters that aren't. You can check that page for an up-to-date list, but some of the most popular are:

Node Exporter
: Exposes hardware and OS metrics exposed by *NIX kernels.

Windows Exporter
: Exposes hardware and OS metrics exposed by Windows.

JMX Exporter
: Scrapes and expose MBeans of a JMX target.

PostgreSQL Exporter
: Retrieves and exposes PostgreSQL server metrics.

Redis Exporter
: Retrieves and exposes Redis server metrics.

Blackbox Exporter
: Allows blackbox probing of endpoints over HTTP, HTTPS, DNS, TCP, or ICMP.

Push Gateway
: A metrics cache that lets ephemeral and batch jobs expose their metrics by pushing them to an intermediary. Technically a core component of Prometheus rather than a distinct exporter.

Obtaining a meter

Before you can instrument an operation, you first have to obtain a `Meter` value from a `MeterProvider`.

As you'll see in "Metric instruments" on page 379, the `metric.Meter` type, which lives in the `go.opentelemetry.io/otel/metric` package, is the means by which all metric collection is configured and reported, either as record batches of synchronous measurements or asynchronous observations.

You can retrieve a `Meter` value as follows:

```go
meter := otel.GetMeterProvider().Meter("fibonacci")
```

You may have noticed that snippet looks almost exactly like the expression used to get a `Tracer` back in "Obtaining a tracer" on page 356. In fact, `otel.GetMeterProvider` is exactly equivalent to `otel.GetTracerProvider`, and works pretty much the same way.

The `otel.GetMeterProvider` function returns the registered global meter provider. If none is registered then a default meter provider is returned that forwards the `Meter` interface to the first registered `Meter` value.

The provider's `Meter` method returns an instance of the `metric.Meter` type. It accepts a string parameter representing the instrumentation name, which by convention is named after the library or package it's instrumenting.

Metric instruments

Once you have a `Meter`, you can create *instruments*, which you can use to make measurements and to instrument your code. However, just as there are several different types of metrics, there are several types of instruments. The type of instrument you use will depend on the type of measurement you're making.

All told, there are 12 *kinds* of instruments available, each with some combination of *synchronicity*, *accumulation* behavior, and data type.

The first of these properties, *synchronicity*, determines how an instrument collects and transmits data:

- *Synchronous instruments* are explicitly called by the user to record a metric, as we'll see in "Synchronous instruments" on page 381.
- *Asynchronous instruments*, also called *observers*, can monitor a specific property and are asynchronously called by the SDK during collection. We'll demonstrate in "Asynchronous instruments" on page 383.

Second, each instrument has an *accumulation* behavior that describes how it tracks the acquisition of new data:

- *Additive* instruments are used to track a sum that can go arbitrarily up or down, like a gauge. They're typically used for measured values like temperatures or current memory usage, but also "counts" that can go up and down, like the number of concurrent requests.
- *Additive monotonic* instruments track monotonically increasing values that can only increase (or be reset to zero on restart), like a counter. Additive monotonic values are often used for metrics like the number of requests served, tasks completed, or errors.
- *Grouping* instruments are intended for capturing a distribution, like a histogram. A grouping instrument samples observations (usually things like request durations or response sizes) and counts them in configurable buckets. It also provides a sum of all observed values.

Finally, each of the previous six kinds of instruments has types that support either `float64` or `int64` input values, for a total of 12 kinds of instruments. Each has an associated type in the `go.opentelemetry.io/otel/metric` package, summarized in Table 11-1.

Table 11-1. The 12 kinds of OpenTelemetry metric instruments, by synchronicity and accumulation behavior.

	Synchronous	Asynchronous
Additive	`Float64UpDownCounter`, `Int64UpDownCounter`	`Float64UpDownSumObserver`, `Int64UpDownSumObserver`
Additive, Monotonic	`Float64Counter`, `Int64Counter`	`Float64SumObserver`, `Int64SumObserver`
Grouping	`Float64ValueRecorder`, `Int64ValueRecorder`	`Float64ValueObserver`, `Int64ValueObserver`

Each of the 12 types has an associated constructor function on the `metric.Meter` type, all with a similar signature. For example, the `NewInt64Counter` method looks like the following:

```go
func (m Meter) NewInt64Counter(name string, options ...InstrumentOption)
    (Int64Counter, error)
```

All 12 constructor methods accept the name of the metric as a `string`, and zero or more `metric.InstrumentOption` values, just like the `NewInt64Counter` method. Similarly, each returns an instrument value of the appropriate type with the given name and options, and can return an error if the name is empty or otherwise invalid, or if the instrument is duplicate registered.

For example, a function that uses the `NewInt64Counter` method to get a new `metric.Int64Counter` from a `metric.Meter` value looks something like the following:

```go
// The requests counter instrument. As a synchronous instrument,
// we'll need to keep it so we can use it later to record data.
var requests metric.Int64Counter

func buildRequestsCounter() error {
    var err error

    // Retrieve the meter from the meter provider.
    meter := otel.GetMeterProvider().Meter(serviceName)

    // Get an Int64Counter for a metric called "fibonacci_requests_total".
    requests, err = meter.NewInt64Counter("fibonacci_requests_total",
        metric.WithDescription("Total number of Fibonacci requests."),
    )

    return err
}
```

Note how we retain a reference to the instrument in the form of the `requests` global variable. For reasons I'll discuss shortly, this is generally specific to synchronous instruments.

But while the metric.Int64Counter happens to be a synchronous instrument, the takeaway here is that synchronous and asynchronous instruments are both obtained in the same way: via the corresponding Metric constructor method. How they're used, however, differs significantly, as we'll see in the subsequent sections.

Synchronous instruments. The initial steps to using a synchronous instrument—retrieving a meter from the meter provider and creating an instrument—are largely the same for both synchronous and asynchronous instruments. We saw these in the previous section.

However, using synchronous instruments differs from using asynchronous instruments in that they're explicitly exercised in your code logic when recording a metric, which means you have to be able to refer to your instrument after it's been created. That's why the above example uses a global requests variable.

Perhaps the most common application is to record individual events by incrementing a counter when an event occurs. The additive instruments even have an Add method for this. The following example uses the requests value that we created in the previous example by adding a call to requests.Add to the API's Fibonacci function that was originally defined in "The Fibonacci service API" on page 363:

```go
// Define our labels here so that we can easily reuse them.
var labels = []label.KeyValue{
    label.Key("application").String(serviceName),
    label.Key("container_id").String(os.Getenv("HOSTNAME")),
}

func Fibonacci(ctx context.Context, n int) chan int {
    // Use the Add method on out metric.Int64Counter instance
    // to increment the counter value.
    requests.Add(ctx, 1, labels...)

    // The rest of the function...
}
```

As you can see, the requests.Add method—which is safe for concurrent use—accepts three parameters:

- The first parameter is the current context in the form of a context.Context value. This is common for all of the synchronous instrument methods.
- The second parameter is the number to increment by. In this case, each call to Fibonacci increases the call counter by one.
- The third parameter is zero or more label.KeyValue values that represent the labels to associate with the data points. This increases the cardinality of the metrics, which, as discussed in "Cardinality" on page 370, is incredibly useful.

Data labels are a powerful tool that allow you to describe data beyond which service or instance emitted it. They can allow you to ask questions of your data that you hadn't thought of before.

It's also possible to group multiple metrics and report them as a batch. This works slightly differently than the `Add` method you saw in the previous example, though. Specifically, for each metric in the batch, you need to:

1. Collect the value or values you want to record.
2. Pass each value to its appropriate instrument's `Measurement` method, which returns a `metric.Measurement` value that wraps your metric and provides some supporting metadata.
3. Pass all of the `metric.Measurement` values to the `meter.RecordBatch`, which atomically records the entire batch of measurements.

These steps are demonstrated in the following example, in which we use the `runtime` package to retrieve two values—the amount of memory and the number of goroutines used by the process—and emit them to the metrics collector:

```go
func updateMetrics(ctx context.Context) {
    // Retrieve the meter from the meter provider.
    meter := otel.GetMeterProvider().Meter(serviceName)

    // Create the instruments that we'll use to report memory
    // and goroutine values. Error values ignored for brevity.
    mem, _ := meter.NewInt64UpDownCounter("memory_usage_bytes",
        metric.WithDescription("Amount of memory used."),
    )
    goroutines, _ := meter.NewInt64UpDownCounter("num_goroutines",
        metric.WithDescription("Number of running goroutines."),
    )

    var m runtime.MemStats

    for {
        runtime.ReadMemStats(&m)

        // Report the values to the instruments, and receive
        // metric.Measurement values in return.
        mMem := mem.Measurement(int64(m.Sys))
        mGoroutines := goroutines.Measurement(int64(runtime.NumGoroutine()))

        // Provide the measurements (and teh context and
        // labels) to the meter.
        meter.RecordBatch(ctx, labels, mMem, mGoroutines)
```

```
            time.Sleep(5 * time.Second)
        }
    }
```

When run as a goroutine, the `updateMetrics` function executes in two parts: an initial setup, and an infinite loop in which it generates and records measurements.

In the set-up phase, it retrieves the `Meter`, defines some metric labels, and creates the instruments. All of these values are created exactly once and are reused in the loop. Note that in addition to types, the instruments are created with names and descriptions indicating the metrics they're instrumenting.

Inside the loop, we first use the `runtime.ReadMemStats` and `runtime.NumGoroutine` functions to retrieve the metrics we want to record (the amount of memory used and the number of running goroutines, respectively). With those values, we use the instruments' `Measurement` methods to generate `metrics.Measurement` values for each metric.

With our `Measurement` values in hand, we pass them into the `meter.RecordBatch` method—which also accepts the current `context.Context` and any labels that we want to attach to the metrics—to officially record them.

Asynchronous instruments. Asynchronous instruments, or *observers*, are created and configured during setup to measure a particular property, and are subsequently called by the SDK during collection. This is especially useful when you have a value you want to monitor without managing your own background recording process.

Just like synchronous instruments, asynchronous instruments are created from a constructor method attached to a `metric.Meter` instance. In total, there are six such functions: a `float64` and `int64` version for each of the three accumulation behaviors. All six have a very similar signature, of which the following is representative:

```
func (m Meter) NewInt64UpDownSumObserver(name string,
    callback Int64ObserverFunc, opts ...InstrumentOption)
    (Int64UpDownSumObserver, error)
```

As you can see, the `NewInt64UpDownSumObserver` accepts the name of the metric as a `string`, something called a `Int64ObserverFunc`, and zero or more instrument options (such as the metric description). Although it returns the observer value, this isn't actually used all that often, though it can return a non-`nil` error if the name is empty, duplicate registered, or otherwise invalid.

The second parameter—the *callback function*—is the heart of any asynchronous instrument. Callback functions are asynchronously called by the SDK upon data collection. There are two kinds, one each for `int64` and `float64`, but they look, feel, and work essentially the same:

```
type Int64ObserverFunc func(context.Context, metric.Int64ObserverResult)
```

When called by the SDK, the callback functions receive the current `context.Context`, and either a `metric.Float64ObserverResult` (for `float64` observers) or `metric.Int64ObserverResult` (for `int64` observers). Both result types have an `Observe` method, which you use to report your results.

This is a lot of little details, but they come together fairly seamlessly. The following function does exactly that, defining two observers:

```go
func buildRuntimeObservers() {
    meter := otel.GetMeterProvider().Meter(serviceName)
    m := runtime.MemStats{}

    meter.NewInt64UpDownSumObserver("memory_usage_bytes",
        func(_ context.Context, result metric.Int64ObserverResult) {
            runtime.ReadMemStats(&m)
            result.Observe(int64(m.Sys), labels...)
        },
        metric.WithDescription("Amount of memory used."),
    )

    meter.NewInt64UpDownSumObserver("num_goroutines",
        func(_ context.Context, result metric.Int64ObserverResult) {
            result.Observe(int64(runtime.NumGoroutine()), labels...)
        },
        metric.WithDescription("Number of running goroutines."),
    )
}
```

When called by `main`, the `buildRuntimeObservers` function defines two asynchronous instruments—`memory_usage_bytes` and `num_goroutines`—each with a callback function that works exactly like the data collection in the `updateMetrics` function that we defined in "Synchronous instruments" on page 381.

In `updateMetrics`, however, we used an infinite loop to synchronously report data. As you can see, using an asynchronous approach for non-event data is not only less work to set up and manage, but has fewer moving parts to worry about later, since there isn't anything else to do once the observers (and their callback functions) are defined and the SDK takes over.

Putting It All Together: Metrics

Now that we have an idea what metrics we're going to collect and how, we can use them to extend the Fibonacci web service that we put together in "Putting It All Together: Tracing" on page 363.

The functionality of the service will remain unchanged. As before, it will be able to accept an `HTTP GET` request, in which the *n*th Fibonacci number can be requested using parameter `n` on the `GET` query string. For example, to request the sixth

Fibonacci number, you should be able to `curl` the service as: `http://localhost:3000?n=6`.

The specific changes we'll be making, and the metrics that we'll be collecting, are as follows:

- Synchronously recording the API request count by adding the `buildRequestsCounter` function to `main` and instrumenting the `Fibonacci` function in the service API as we described in "Synchronous instruments" on page 381
- Asynchronously recording the processes' memory used, and number of active goroutines, by adding the `buildRuntimeObservers` described in "Asynchronous instruments" on page 383 to the `main` function

Starting your services

Once again, start your service by running its main function:

```
$ go run .
```

As before, your terminal should pause. You can stop the service with a Ctrl-C.

Next, you'll start the Prometheus server. But before you do, you'll need to create a minimal configuration file for it. Prometheus has a ton of available configuration options, but the following should be perfectly sufficient. Copy and paste it into a file named `prometheus.yml`:

```
scrape_configs:
- job_name: fibonacci
  scrape_interval: 5s
  static_configs:
  - targets: ['host.docker.internal:3000']
```

This configuration defines a single target named `fibonacci` that lives at `host.docker.internal:3000` and will be scraped every five seconds (down from the default of every minute).

What is host.docker.internal?

The name `host.docker.internal` is a special DNS name defined in Docker Desktop for both Mac and Windows that resolves to the internal IP address used by the host, allowing the container to interact with host processes.

Importantly, this address is provided for development convenience and won't work in a production environment outside of Docker Desktop for Mac and Windows (i.e., it's not supported by default in Docker on Linux).

Once you've created the file `prometheus.yml`, you can start Prometheus. The easiest way to do this is a container using Docker:

```
docker run -d --name prometheus                                    \
  -p 9090:9090                                                     \
  -v "${PWD}/prometheus.yml:/etc/prometheus/prometheus.yml" \
  prom/prometheus:v2.23.0
```

If you're using Linux for development, you'll need to add the parameter `--add-host=host.docker.internal:host-gateway` to the above command. *But do not use this in production.*

Now that your services are both running, you can send a request to the service:

```
$ curl localhost:3000?n=6
13
```

Behind the scenes, OpenTelemetry has just recorded a value for the number of requests (recursive and otherwise) made to its `Fibonacci` function.

Metric endpoint output

Now that your service is running, you can always examine its exposed metrics directly with a standard curl to its `/metrics` endpoint:

```
$ curl localhost:3000/metrics
# HELP fibonacci_requests_total Total number of Fibonacci requests.
# TYPE fibonacci_requests_total counter
fibonacci_requests_total{application="fibonacci",container_id="d35f0bef2ca0"} 25
# HELP memory_usage_bytes Amount of memory used.
# TYPE memory_usage_bytes gauge
memory_usage_bytes{application="fibonacci",container_id="d35f0bef2ca0"}
  7.5056128e+07
# HELP num_goroutines Number of running goroutines.
# TYPE num_goroutines gauge
num_goroutines{application="fibonacci",container_id="d35f0bef2ca0"} 6
```

As you can see, all three of the metrics you're recording—as well as their types, descriptions, labels, and values—are listed here. Don't be confused if the value of `container_id` is empty: that just means you're not running in a container.

Viewing your results in Prometheus

Now that you've started your service, started Prometheus, and run a query or two to the service to seed some data, it's time to visualize your work in Prometheus. Again, Prometheus isn't a full-fledged graphing solution (you'll want to use something like Grafana for that), but it does offer a simple interface for executing arbitrary queries.

You can access this interface by browsing to `localhost:9090`. You should be presented with a minimalist interface with a search field. To see the value of your metric over time, enter its name in the search field, hit enter, and click the "graph" tab. You should be presented with something like the screenshot in Figure 11-6.

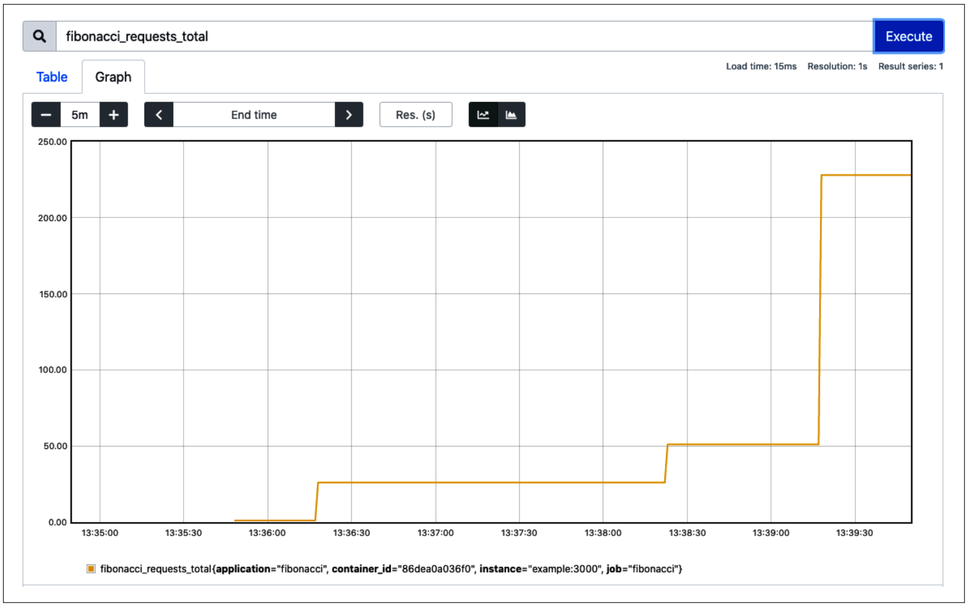


Figure 11-6. Screenshot of the Prometheus interface, displaying the value of the `fibonacci_requests_total` metric after three calls to the Fibonacci service

Now that you're collecting data, take a moment to run a few more queries and see how the graph changes. Maybe even look at some other metrics. Enjoy!

Logging

A *log* is an immutable record of *events*—discrete occurrences that are worth recording—emitted by an application over time. Traditionally, logs were stored as append-only files, but these days, a log is just as likely to take the form of some kind of searchable data store.

So, what's there to say about logging, other than that it's a really good idea that's been around as long as electronic computing has? It's the OG of observability methods.

There's actually quite a bit to say, largely because it's really, really easy to do logging in a way that makes your life harder than it needs to be.

Of the Three Pillars of Observability, logs are by far the easiest to generate. Since there's no initial processing involved in outputting a log event, in its simplest form it's as easy as adding a `print` statement to your code. This makes logs really good at providing lots and lots of context-rich data about what a component is doing or experiencing.

But this free-form aspect to logging cuts both ways. While it's possible (and often tempting) to output whatever you think might be useful, the verbose, unstructured logs are difficult to extract usable information from, especially at scale. To get the most out of logging, events should be structured, and that structure doesn't come for free. It has to be intentionally considered and implemented.

Another, particularly underappreciated, pitfall of logging is that generating lots of events puts significant pressure on disk and/or network I/O. It's not unusual for half or more of available bandwidth to be consumed this way. What's more, this pressure tends to scale linearly with load: `N` users each doing `M` things translates to `N*M` log events being emitted, with potentially disastrous consequences for scalability.

Finally, for logs to be meaningfully useful, they have to be processed and stored in a way that makes them accessible. Anybody who's ever had to manage logs at scale can tell you that it's notoriously operationally burdensome to self-manage and self-host, and absurdly expensive to have somebody else manage and host.

In the remainder of this section, we'll first discuss some high-level practices for logging at scale, followed by how to implement them in Go.

Better Logging Practices

As simple as the act of logging may seem on the face of it, it's also really easy to log in a way that makes life harder for you and anybody who has to use your logs after you. Awkward logging issues, like having to navigate unstructured logs or higher-than-expected resource consumption, which are annoying in small deployments, become major roadblocks at scale.

As you'll see, for this reason and others, the best practices around logging tend to focus on maximizing the quality, and minimizing the quantity, of logging data generated and retained.

It goes without saying that you shouldn't log sensitive business data or personally identifiable information.

Treat logs as streams of events

How many times have you looked at log output and been confronted with an inscrutable stream of consciousness? How useful was it? Better than nothing, maybe, but probably not by much.

Logs shouldn't be treated as data sinks to be written to and forgotten until something is literally on fire, and they definitely shouldn't be a garbage dump where you send random thoughts and observations.

Instead, as we saw back in Chapter 6, logs should be treated as a *stream of events*, and should be written, unbuffered, directly to `stdout` and `stderr`. Though seemingly simple (and perhaps somewhat counterintuitive), this small change in perspective provides a great deal of freedom.

By moving the responsibility for log management out of the application code, it's freed from concerns about implementation trivialities like routing or storage of its log events, allowing the executor to decide what happens to them.

This approach provides quite a lot of freedom for how you manage and consume your logs. In development, you can keep an eye on your service's behavior by sending them directly to a local terminal. In production, the execution environment can capture and redirect log events to a log indexing system like ELK or Splunk for review and analysis, or perhaps a data warehouse for long-term storage.

Treat logs as streams of events, and write each event, unbuffered, directly to `stdout` and `stderr`.

Structure events for parsing

Logging, in its simplest and most primitive form, is technically possible using nothing more than `fmt.Println` statements. The result, however, would be a set of unformatted strings of questionable utility.

Fortunately, it's more common for programmers to use Go's standard `log` library, which is conveniently located and easy to use, and generates helpful timestamps. But how useful would a terabyte or so of log events formatted like the following be?

```
2020/11/09 02:15:10AM User 12345: GET /help in 23ms
2020/11/09 02:15:11AM Database error: connection reset by peer
```

Certainly, it's better than nothing, but you're still confronted with a mostly unstructured string, albeit an unstructured string with a timestamp. You still have to parse the arbitrary text to extract the meaningful bits.

Compare that to the equivalent messages outputted by a structured logger:[13]

```
{"time":1604888110, "level":"info", "method":"GET", "path":"/help",
        "duration":23, "message":"Access"}
{"time":1604888111, "level":"error", "error":"connection reset by peer",
        "database":"user", "message":"Database error"}
```

The above log structure places all of the key elements into properties of a JavaScript object, each with:

`time`
: A timestamp, which is a piece of contextual information that's critical for tracking and correlating issues. Note that the JSON example is also in an easily-parsable format that's far less computationally expensive to extract meaning from than the first, barely structured example. When you're processing billions of log events, little things add up.

`level`
: A log level, which is a label that indicates the level of importance for the log event. Frequently used levels include `INFO`, `WARN`, and `ERROR`. These are also key for filtering out low-priority messages that might not be relevant in production.

One or more contextual elements
: These contain background information that provides insight into the state of the application at the time of the message. The *entire point* of a log event is to express this context information.

In short, the structured log form is easier, faster, and cheaper to extract meaning from, and the results are far easier to search, filter, and aggregate.

Structure your logs for parsing by computers, not for reading by humans.

Less is (way) more

Logging isn't free. In fact, it's very expensive.

Imagine you have a service deployed to a server running in AWS. Nothing fancy, just a standard server with a standard, general-purpose disk capable of a sustained throughput of 16 MiB/second.

Let's say that your service likes to be thorough, so it fastidiously logs events acknowledging each request, response, database call, calculation status, and various other bits of information, totaling sixteen 1024-byte events for each request the service handles. It's a little verbose, but nothing too unusual so far.

13 Any wrapping in the example is for the benefit of formatting for presentation only. Don't use line breaks in your log events if you can help it.

But this adds up. In a scenario in which the service handles 512 requests per second—a perfectly reasonable number for a highly concurrent service—your service would produce 8192 events/second. At 16 KiB per event, that's a total of 8 MiB/second of log events, or *half of your disk's I/O capacity*. That's quite a burden.

What if we skip writing to disk and forward events straight to a log-hosting service? Well, the bad news is that we then have to transfer and store our logs, and that gets expensive. If you're sending the data across the internet to a log provider like Splunk or Datadog, you'll have to pay your cloud provider a data transfer fee. For AWS, this amounts to US$0.08/GB, which at an average rate of 8 MiB/s—about 1 TiB every day and a half—comes to almost $250,000/year for a single instance. Fifty such instances would run more than $12 million dollars in data transfer costs alone.

Obviously, this example doesn't take into account fluctuations in load due to hour of day or day of week. But it clearly illustrates that logging can get very expensive very quickly, so log only what's useful, and be sure to limit log generation in production by using severity thresholds. A "warning" threshold is common.

Dynamic sampling

Because the kind of events that are produced by debug events tend to be both high-volume and low-fidelity, it's pretty standard practice to eliminate them from production output by setting the log level to `WARNING`. But debug logs aren't *worthless*, are they?[14] As it turns out, they become really useful really fast when you're trying to chase down the root cause of an outage, which means you have to waste precious incident time turning debug logs on just long enough for you find the problem. Oh, and don't forget to turn them off afterwards.

However, by *dynamically sampling* your logs—recording some proportion of events and dropping the rest—you can still have your debug logs—but not too many—available in production, which can help drive down the time to recovery during an incident.

Having some debug logs in production can be *really* useful when things are on fire.

Logging with Go's Standard log Package

Go includes a standard logging package, appropriately named `log`, that provides some basic logging features. While it's very bare bones, it still has just about everything you need to put together a basic logging strategy.

Besides importing the `log` package, using it doesn't require any kind of setup.

14 If they are, why are you producing them at all?

Its most basic functions can be leveraged with a selection of functions very similar to the various `fmt` print functions you may be familiar with:

```
func Print(v ...interface{})
func Printf(format string, v ...interface{})
func Println(v ...interface{})
```

You may have noticed what is perhaps the most glaring omission from the `log` package: that it doesn't support logging levels. However, what it lacks in functionality, it makes up for in simplicity and ease of use.

Here's the most basic logging example:

```
package main

import "log"

func main() {
    log.Print("Hello, World!")
}
```

When run, it provides the following output:

```
$ go run .
2020/11/10 09:15:39 Hello, World!
```

As you can see, the `log.Print` function—like all of the `log` logging functions—adds a timestamp to its messages without any additional configuration.

The special logging functions

Although `log` sadly doesn't support log levels, it does offer some other interesting features. Namely, a class of convenience functions that couple outputting log events with another useful action.

The first of these is the `log.Fatal` functions. There are three of these, each corresponding to a different `log.PrintX` function, and each equivalent to calling its corresponding print function followed by a call to `os.Exit(1)`:

```
func Fatal(v ...interface{})
func Fatalf(format string, v ...interface{})
func Fatalln(v ...interface{})
```

Similarly, `log` offers a series of `log.Panic` functions, which are equivalent to calling its corresponding `log.PrintX` followed by a call to `panic`:

```
func Panic(v ...interface{})
func Panicf(format string, v ...interface{})
func Panicln(v ...interface{})
```

Both of these sets of functions are useful, but they're not used nearly as often as the `log.Print` functions, typically in error handling where it makes sense to report the error and halt.

Logging to a custom writer

By default, the `log` package prints to `stderr`, but what if you want to redirect that output elsewhere? The `log.SetOutput` function allows you to do exactly that by letting you specify a custom `io.Writer` to write to.

This allows you to, for example, send your logs to a file if you want to. As we mention in "Less is (way) more" on page 390, writing logs to files generally isn't advisable, but it can be useful under certain circumstances.

This is demonstrated in the following using `os.OpenFile` to open the target file, and using `log.SetOutput` to define it as the log writer:

```go
package main

import (
    "log"
    "os"
)

func main() {
    // O_APPEND = Append data to the file when writing
    // O_CREATE = Create a new file if none exists
    // O_WRONLY = Open the file write-only
    flags := os.O_APPEND | os.O_CREATE | os.O_WRONLY

    file, err := os.OpenFile("log.txt", flags, 0666)
    if err != nil {
        log.Fatal(err)
    }

    log.SetOutput(file)

    log.Println("Hello, World!")
}
```

When run, the following is written to the file `log.txt`:

```
$ go run .; tail log.txt
2020/11/10 09:17:05 Hello, World!
```

The fact that `log.SetOutput` accepts an interface means that a wide variety of destinations can be supported just by satisfying the `io.Writer` contract. You could even, if you so desired, create an `io.Writer` implementation that forwards to a log processor like Logstash or a message broker like Kafka. The possibilities are unlimited.

Log flags

The `log` package also allows you to use constants to enrich log messages with additional context information, such as the filename, line number, date, and time.

For example, adding the following line to our above “Hello, World”:

```
log.SetFlags(log.Ldate | log.Ltime | log.Lshortfile)
```

Will result in a log output like the following:

```
2020/11/10 10:14:36 main.go:7: Hello, World!
```

As you can see, it includes the date in the local time zone (`log.Ldate`), the time in the local time zone (`log.Ltime`), and the final file name element and line number of the `log` call (`log.Lshortfile`).

We don’t get any say over the order in which the log parts appear or the format in which they are presented, but if you want that kind of flexibility, you probably want to use another logging framework, such as Zap.

The Zap Logging Package

Of the Three Pillars of Observability, logging is the one that’s least supported by OpenTelemetry. Which is to say that it isn’t supported at all, at least at the time of this writing (though it will be incorporated in time).

So, for now, rather than discuss the OpenTelemetry Logging API, we’ll cover another excellent library: Zap, a JSON-formatted logger designed to allocate memory as infrequently as possible, and to use reflection and string formatting as little as possible.

Zap is currently one of the two most popular Go logging packages, alongside Logrus. Logrus is actually a little more popular, but three main factors drove me to choose Zap for this book instead. First, Zap is known for its speed and low memory impact (which is useful at scale). Second, it has a “structured first” philosophy which, as I asserted in “Structure events for parsing” on page 389, is incredibly desirable. Finally, Logrus is now in maintenance mode, and isn’t introducing any new features.

How fast is Zap, exactly? It’s really fast. For a minimalist example, Table 11-2 shows comparisons of benchmarks between several common structured logging packages, without including any context or `printf`-style templating.

Table 11-2. Relative benchmarks of structured logging packages for a message with no context or `printf`-style templating.

Package	Time	Time % to Zap	Objects Allocated
Zap	118 ns/op	+0%	0 allocs/op
Zap (sugared)	191 ns/op	+62%	2 allocs/op
Zerolog	93 ns/op	-21%	0 allocs/op
Go-kit	280 ns/op	+137%	11 allocs/op
Standard library	499 ns/op	+323%	2 allocs/op
Logrus	3129 ns/op	+2552%	24 allocs/op
Log15	3887 ns/op	+3194%	23 allocs/op

These numbers were developed using Zap's own benchmarking suite, but I did examine, update, and execute the benchmarks myself. Of course, as with any benchmarking, take these numbers with a grain of salt. The two standouts here are Go's own standard `log` library, which had a runtime about triple Zap's standard logger, and Logrus, which took a very significant 25 times Zap's time.

But we're supposed to use context fields, aren't we? What does Zap look like then? Well, those results are even more striking.

Table 11-3. Relative benchmarks of structured logging packages for a message with 10 context fields.

Package	Time	Time % to Zap	Objects Allocated
Zap	862 ns/op	+0%	5 allocs/op
Zap (sugared)	1250 ns/op	+45%	11 allocs/op
Zerolog	4021 ns/op	+366%	76 allocs/op
Go-kit	4542 ns/op	+427%	105 allocs/op
Logrus	29501 ns/op	+3322%	125 allocs/op
Log15	29906 ns/op	+3369%	122 allocs/op

Zap's lead over Logrus has extended to a (very impressive) factor of 33X; the standard `log` library isn't included in this table because it doesn't even support context fields.

Alright then, so how do we use it?

Creating a Zap logger

The first step to logging with Zap is to create a `zap.Logger` value.

Of course, before you do that, you first need to import the Zap package, as follows:

```go
import "go.uber.org/zap"
```

Once you've imported Zap, you can build your `zap.Logger` instance. Zap allows you to configure several aspects of your logging behavior, but the most straightforward way to build a `zap.Logger` is to use Zap's opinionated preset constructor functions—`zap.NewExample`, `zap.NewProduction`, and `zap.NewDevelopment`—each of which build a logger via a single function call:

```
logger, err := zap.NewProduction()
if err != nil {
    log.Fatalf("can't initialize zap logger: %v", err)
}
```

Typically, this will be done in an `init` function and the `zap.Logger` value maintained globally. Zap loggers are safe for concurrent use.

The three available presets are usually perfectly fine for small projects, but larger projects and organizations may want a bit more customization. Zap provides the `zap.Config` struct for exactly this purpose, and while the specifics are beyond the scope of this book, the Zap documentation describes its use in some detail.

Writing logs with Zap

One of the more unique aspects of Zap is that every logger actually has two easily interchangeable forms—standard and "sugared"—that vary somewhat in efficiency and usability.

The standard `zap.Logger` implementation emphasizes performance and type safety. It's slightly faster than the `SugaredLogger` and allocates far less, but it only supports structured logging which does make it a little more awkward to use:

```
logger, _ := zap.NewProduction()

// Structured context as strongly typed Field values.
logger.Info("failed to fetch URL",
    zap.String("url", url),
    zap.Int("attempt", 3),
    zap.Duration("backoff", time.Second),
)
```

The output of which will look something like the following:

```
{"level":"info", "msg":"failed to fetch URL",
        "url":"http://example.com", "attempt":3, "backoff":"1s"}
```

In contexts where performance is good but not absolutely critical (which is most of the time, probably) you can use the `SugaredLogger`, which is easily obtainable from a standard logger via its `Sugar` method.

The `SugaredLogger` still provides structured logging, but its functions for doing so are loosely typed, as opposed to the standard logger's strong context typing. Despite using runtime reflection behind the scenes, its performance is still very good.

The `SugaredLogger` even includes `printf`-style logging methods, for convenience. (Remember, though, that when it comes to logging, context is king.)

All of these features are demonstrated in the following:

```
logger, _ := zap.NewProduction()
sugar := logger.Sugar()

// Structured context as loosely typed key-value pairs.
sugar.Infow("failed to fetch URL",
    "url", url,
    "attempt", 3,
    "backoff", time.Second,
)

sugar.Infof("failed to fetch URL: %s", url)
```

The output of which will look something like the following:

```
{"level":"info", "msg":"failed to fetch URL",
        "url":"http://example.com", "attempt":3, "backoff":"1s"}
{"level":"info", "msg":"failed to fetch URL: http://example.com"}
```

Don't create a new `Logger` for every function. Instead, create a global instance, or use the `zap.L` or `zap.S` functions to get Zap's global standard or sugared loggers, respectively.

Using dynamic sampling in Zap

You may recall from "Dynamic sampling" on page 391 that dynamic sampling is a technique in which incoming log entries are sampled by capping recorded events to some maximum number per unit of time.

If done broadly, this technique can be used to manage the CPU and I/O load of your logging while preserving a representative subset of events. If targeted to a particular class of otherwise high-volume and low-fidelity events, such as debug logs, dynamic sampling can ensure their availability for production troubleshooting without consuming too much storage.

Zap supports dynamic sampling, which is configurable using the `zap.SamplingConfig` structure, shown here:

```
type SamplingConfig struct {
    // Initial sets the cap on the number of events logged each second.
    Initial    int

    // Thereafter sets the proportion of events that are logged each second
    // after Initial is exceeded. A value of 3 indicates one event in every
    // 3 is logged.
```

```
        Thereafter int

        // Hook (if defined) is called after each "log/no log" decision.
        Hook       func(zapcore.Entry, zapcore.SamplingDecision)
    }
```

Using `zap.SamplingConfig` allows you to define the number of initial events with the same level and message permitted each second (`Initial`), after which only every *n*th message (`Thereafter`) is logged. The rest are dropped.

The following example demonstrates how to build a new `zap.Logger` using a preconfigured `zap.Config` instance:

```
package main

import (
    "fmt"

    "go.uber.org/zap"
    "go.uber.org/zap/zapcore"
)

func init() {
    cfg := zap.NewDevelopmentConfig()
    cfg.EncoderConfig.TimeKey = ""          // Turn off timestamp output

    cfg.Sampling = &zap.SamplingConfig{
        Initial:    3,                      // Allow first 3 events/second
        Thereafter: 3,                      // Allows 1 per 3 thereafter
        Hook: func(e zapcore.Entry, d zapcore.SamplingDecision) {
            if d == zapcore.LogDropped {
                fmt.Println("event dropped...")
            }
        },
    }

    logger, _ := cfg.Build()                // Constructs the new logger

    zap.ReplaceGlobals(logger)              // Replace Zap's global logger
}
```

The above example creates a new `zap.Logger` and sets it as Zap's global logger. It does this in several steps.

First, the example creates a new `zap.Config` struct. For convenience, this example uses the predefined `zap.NewDevelopmentConfig` function, which provides a `zap.Config` value that produces human-readable output and a threshold of `DebugLevel` and above.

If you like, the zap.NewProductionConfig function, which returns a preconfigured zap.Config value with a threshold of InfoLevel and encodes events in JSON. If you really want to, you can even create your own zap.Config from scratch.

Next, the example creates a new zap.SamplingConfig on the zap.Config, which instructs the Zap sampler to keep the first three of any similar events in a given second, and to drop all but every third message thereafter (each second).

The Hook function is invoked after each sampling decision. The example will write a message if it sees that an event has been dropped.

Finally, the example uses the Config's Build method to construct a zap.Logger from the Config, and uses zap.ReplaceGlobals to replace Zap's global Logger. Zap's global logger and sugared logger can be accessed by using the zap.L and zap.S functions, respectively.

But does it work as we expect? Well, let's see:

```go
func main() {
    for i := 1; i <= 10; i++ {
        zap.S().Infow(
            "Testing sampling",
            "index", i,
        )
    }
}
```

The above function logs 10 events, but with our sampling configuration we should see only the first 3 events, and then every third after that (6 and 9). Is that what we see?

```
$ go run .
INFO    zap/main.go:39    Testing sampling    {"index": 1}
INFO    zap/main.go:39    Testing sampling    {"index": 2}
INFO    zap/main.go:39    Testing sampling    {"index": 3}
event dropped...
event dropped...
INFO    zap/main.go:39    Testing sampling    {"index": 6}
event dropped...
event dropped...
INFO    zap/main.go:39    Testing sampling    {"index": 9}
event dropped...
```

The output is exactly as we expected. Clearly, log sampling is a very powerful technique, and, when used properly, can provide significant value.

Summary

There's a lot of hype around observability, and with its promises to dramatically shorten development feedback loops and generally make complexity manageable again, it's easy to see why.

I wrote a little at the start of this chapter about observability and its promises, and a little more about how observability *isn't* done. Unfortunately, *how to do* observability is a really, really big subject, and the limitations of time and space meant that I wasn't able to say as much about that as I certainly would have liked.[15] Fortunately, with some pretty great books on the horizon (most notably *Observability Engineering* by Charity Majors and Liz Fong-Jones (O'Reilly)), that void won't go unfilled for long.

By far, however, most of this chapter was spent talking about the Three Pillars of Observability in turn, specifically how to implement them using OpenTelemetry, where possible.

All told, this was a challenging chapter. Observability is a vast subject about which not that much is written yet, and, as a result of its newness, the same is true of Open-Telemetry. Even its own documentation is limited and spotty in parts. On the plus side, I got to spend a lot of time in the source code.

15 This is a Go book, after all. At least that's what I keep telling my incredibly patient editors.

Fin.