

Chapter 2. Architectures

Distributed systems are often complex pieces of software of which the components are by definition dispersed across multiple machines. To master their complexity, it is crucial that these systems are properly organized. There are different ways on how to view the organization of a distributed system, but an obvious one is to make a distinction between the logical organization of the collection of software components and on the other hand the actual physical realization.

The organization of distributed systems is mostly about the software components that constitute the system. These software architectures tell us how the various software components are to be organized and how they should interact. In this chapter we will first pay attention to some commonly applied approaches toward organizing (distributed) computer systems.

The actual realization of a distributed system requires that we instantiate and place software components on real machines. There are many different choices that can be made in doing so. The final instantiation of a software architecture is also referred to as a system architecture. In this chapter we will look into traditional centralized architectures in which a single server implements most of the software components (and thus functionality), while remote clients can access that server using simple communication means. In addition, we consider decentralized architectures in which machines more or less play equal roles, as well as hybrid organizations.

As we explained in Chap. 1, an important goal of distributed systems is to separate applications from underlying platforms by providing a middleware layer. Adopting such a layer is an important architectural decision, and its main purpose is to provide distribution transparency. However, trade-offs need to be made to achieve transparency, which has led to various techniques to make middleware adaptive. We discuss some of the more commonly applied ones in this chapter, as they affect the organization of the middleware itself.

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Adaptability in distributed systems can also be achieved by having the system monitor its own behavior and taking appropriate measures when needed. This insight has led to a class of what are now referred to as autonomic systems. These distributed systems are frequently organized in the form of feedback control loops, which form an important architectural element during a system's design. In this chapter, we devote a section to autonomic distributed systems.

2.1. Architectural Styles

We start our discussion on architectures by first considering the logical organization of distributed systems into software components, also referred to as software architecture (Bass et al., 2003). Research on software architectures has matured considerably and it is now commonly accepted that designing or adopting an architecture is crucial for the successful development of large systems.

For our discussion, the notion of an architectural style is important. Such a style is formulated in terms of components, the way that components are connected to each other, the data exchanged between components, and finally how these elements are jointly configured into a system. A component is a modular unit with well-defined required and provided interfaces that

is replaceable within its environment (OMG, 2004b). As we shall discuss below, the important issue about a component for distributed systems is that it can be replaced, provided we respect its interfaces. A somewhat more difficult concept to grasp is that of a connector, which is generally described as a mechanism that mediates communication, coordination, or cooperation among components (Mehta et al., 2000; and Shaw and Clements, 1997). For example, a connector can be formed by the facilities for (remote) procedure calls, message passing, or streaming data.

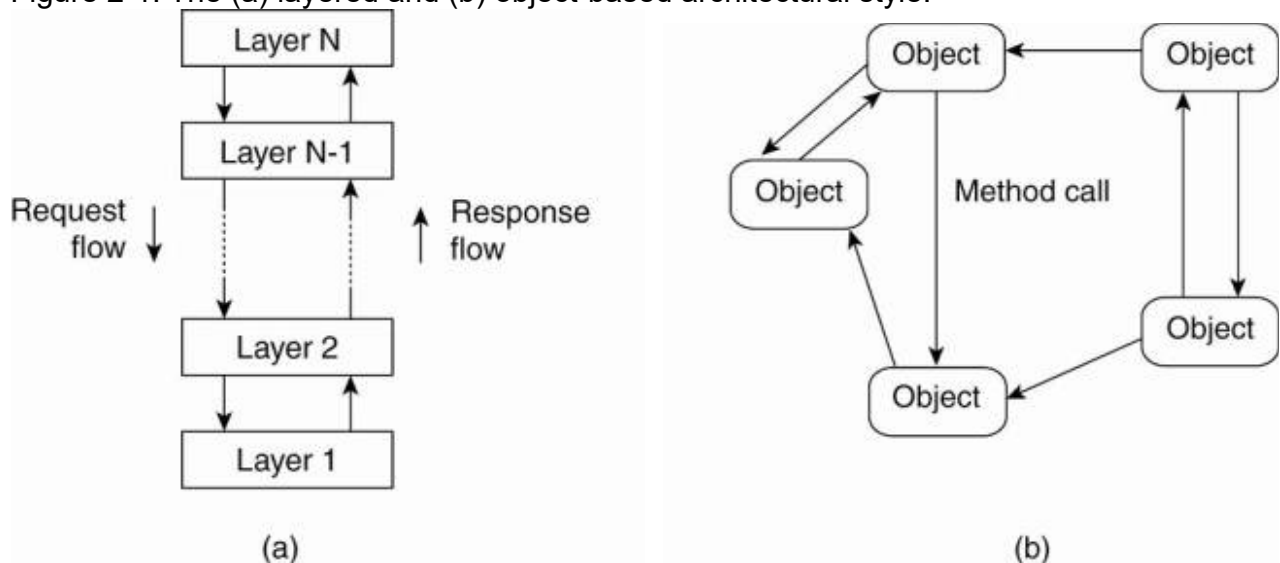
Using components and connectors, we can come to various configurations, which, in turn have been classified into architectural styles. Several styles have by now been identified, of which the most important ones for distributed systems are:

1. Layered architectures
2. Object-based architectures
3. Data-centered architectures
4. Event-based architectures

The basic idea for the layered style is simple: components are organized in a layered fashion where a component at layer L_i is allowed to call components at the underlying layer L_{i-1} , but not the other way around, as shown in Fig. 2-1(a). This model has been widely adopted by the networking community; we briefly review it in Chap. 4. An key observation is that control generally flows from layer to layer: requests go down the hierarchy whereas the results flow upward.

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Figure 2-1. The (a) layered and (b) object-based architectural style.



A far looser organization is followed in object-based architectures, which are illustrated in Fig. 2-1(b). In essence, each object corresponds to what we have defined as a component, and

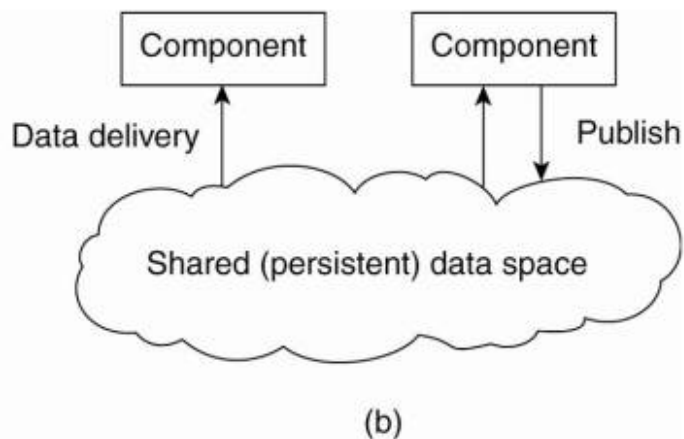
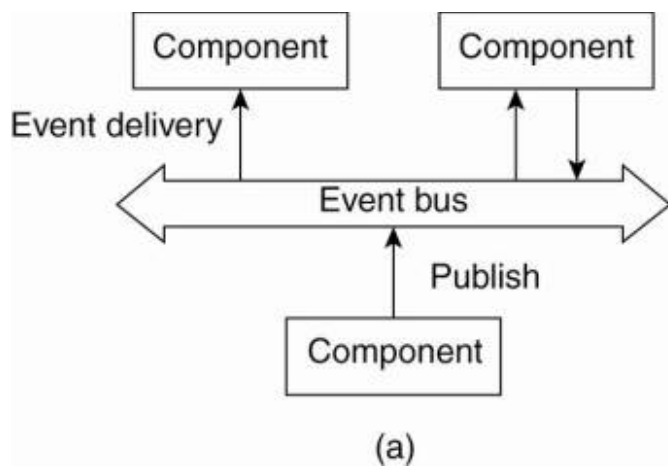
these components are connected through a (remote) procedure call mechanism. Not surprisingly, this software architecture matches the client-server system architecture we described above. The layered and object-based architectures still form the most important styles for large software systems (Bass et al., 2003).

Data-centered architectures evolve around the idea that processes communicate through a common (passive or active) repository. It can be argued that for distributed systems these architectures are as important as the layered and object-based architectures. For example, a wealth of networked applications have been developed that rely on a shared distributed file system in which virtually all communication takes place through files. Likewise, Web-based distributed systems, which we discuss extensively in Chap. 12, are largely data-centric: processes communicate through the use of shared Web-based data services.

In event-based architectures, processes essentially communicate through the propagation of events, which optionally also carry data, as shown in Fig. 2-2(a). For distributed systems, event propagation has generally been associated with what are known as publish/subscribe systems (Eugster et al., 2003). The basic idea is that processes publish events after which the middleware ensures that only those processes that subscribed to those events will receive them. The main advantage of event-based systems is that processes are loosely coupled. In principle, they need not explicitly refer to each other. This is also referred to as being decoupled in space, or referentially decoupled.

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Figure 2-2. The (a) event-based and (b) shared data-space architectural style.



Event-based architectures can be combined with data-centered architectures, yielding what is also known as shared data spaces. The essence of shared data spaces is that processes are now also decoupled in time: they need not both be active when communication takes place. Furthermore, many shared data spaces use a SQL-like interface to the shared repository in that sense that data can be accessed using a description rather than an explicit reference, as is the case with files. We devote Chap. 13 to this architectural style.

What makes these software architectures important for distributed systems is that they all aim at achieving (at a reasonable level) distribution transparency. However, as we have argued, distribution transparency requires making trade-offs between performance, fault tolerance, ease-of-programming, and so on. As there is no single solution that will meet the requirements for all possible distributed applications, researchers have abandoned the idea that a single distributed system can be used to cover 90% of all possible cases.

2.2. System Architectures

Now that we have briefly discussed some common architectural styles, let us take a look at how many distributed systems are actually organized by considering where software components are placed. Deciding on software components, their interaction, and their placement leads to an instance of a software architecture, also called a system architecture

(Bass et al., 2003). We will discuss centralized and decentralized organizations, as well as various hybrid forms.

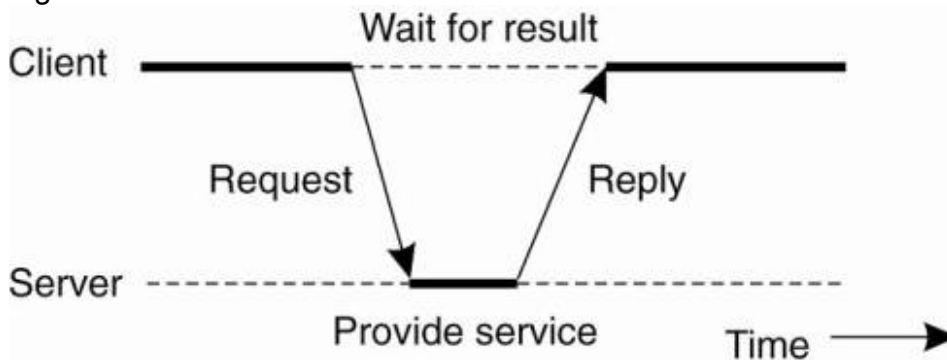
2.2.1. Centralized Architectures

Despite the lack of consensus on many distributed systems issues, there is one issue that many researchers and practitioners agree upon: thinking in terms of clients that request services from servers helps us understand and manage the complexity of distributed systems and that is a good thing.

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In the basic client-server model, processes in a distributed system are divided into two (possibly overlapping) groups. A server is a process implementing a specific service, for example, a file system service or a database service. A client is a process that requests a service from a server by sending it a request and subsequently waiting for the server's reply. This client-server interaction, also known as request-reply behavior is shown in Fig. 2-3

Figure 2-3. General interaction between a client and a server.



Communication between a client and a server can be implemented by means of a simple connectionless protocol when the underlying network is fairly reliable as in many local-area networks. In these cases, when a client requests a service, it simply packages a message for the server, identifying the service it wants, along with the necessary input data. The message is then sent to the server. The latter, in turn, will always wait for an incoming request, subsequently process it, and package the results in a reply message that is then sent to the client.

Using a connectionless protocol has the obvious advantage of being efficient. As long as messages do not get lost or corrupted, the request/reply protocol just sketched works fine. Unfortunately, making the protocol resistant to occasional transmission failures is not trivial. The only thing we can do is possibly let the client resend the request when no reply message comes in. The problem, however, is that the client cannot detect whether the original request message was lost, or that transmission of the reply failed. If the reply was lost, then resending a request may result in performing the operation twice. If the operation was something like "transfer \$10,000 from my bank account," then clearly, it would have been better that we simply reported an error instead. On the other hand, if the operation was "tell me how much money I

have left," it would be perfectly acceptable to resend the request. When an operation can be repeated multiple times without harm, it is said to be idempotent. Since some requests are idempotent and others are not it should be clear that there is no single solution for dealing with lost messages. We defer a detailed discussion on handling transmission failures to Chap. 8.

As an alternative, many client-server systems use a reliable connection-oriented protocol. Although this solution is not entirely appropriate in a local-area network due to relatively low performance, it works perfectly fine in wide-area systems in which communication is inherently unreliable. For example, virtually all Internet application protocols are based on reliable TCP/IP connections. In this case, whenever a client requests a service, it first sets up a connection to the server before sending the request. The server generally uses that same connection to send the reply message, after which the connection is torn down. The trouble is that setting up and tearing down a connection is relatively costly, especially when the request and reply messages are small.

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Application Layering

The client-server model has been subject to many debates and controversies over the years. One of the main issues was how to draw a clear distinction between a client and a server. Not surprisingly, there is often no clear distinction. For example, a server for a distributed database may continuously act as a client because it is forwarding requests to different file servers responsible for implementing the database tables. In such a case, the database server itself essentially does no more than process queries.

However, considering that many client-server applications are targeted toward supporting user access to databases, many people have advocated a distinction between the following three levels, essentially following the layered architectural style we discussed previously:

1. The user-interface level
2. The processing level
3. The data level

The user-interface level contains all that is necessary to directly interface with the user, such as display management. The processing level typically contains the applications. The data level manages the actual data that is being acted on.

Clients typically implement the user-interface level. This level consists of the programs that allow end users to interact with applications. There is a considerable difference in how sophisticated user-interface programs are.

The simplest user-interface program is nothing more than a character-based screen. Such an interface has been typically used in mainframe environments. In those cases where the mainframe controls all interaction, including the keyboard and monitor, one can hardly speak of a client-server environment. However, in many cases, the user's terminal does some local processing such as echoing typed keystrokes, or supporting form-like interfaces in which a complete entry is to be edited before sending it to the main computer.

Nowadays, even in mainframe environments, we see more advanced user interfaces. Typically, the client machine offers at least a graphical display in which pop-up or pull-down menus are used, and of which many of the screen controls are handled through a mouse instead of the keyboard. Typical examples of such interfaces include the X-Windows interfaces as used in many UNIX environments, and earlier interfaces developed for MS-DOS PCs and Apple Macintoshes.

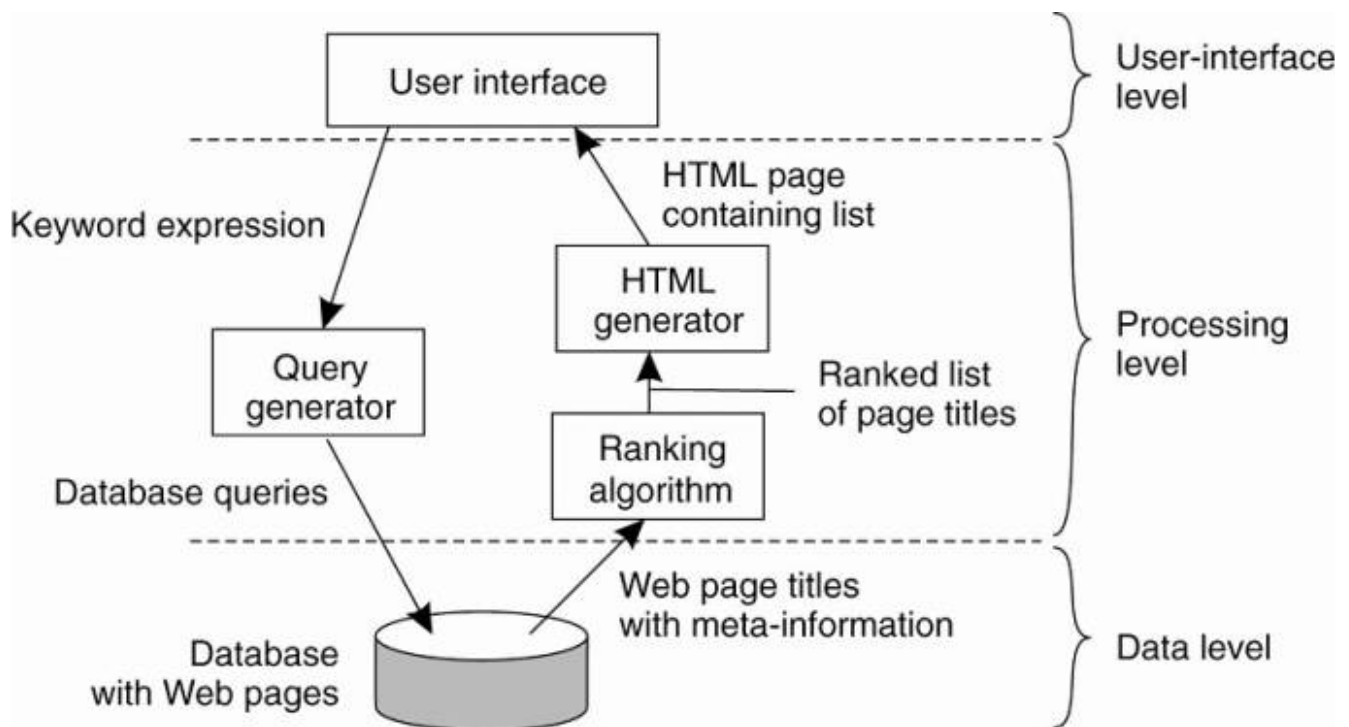
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Modern user interfaces offer considerably more functionality by allowing applications to share a single graphical window, and to use that window to exchange data through user actions. For example, to delete a file, it is usually possible to move the icon representing that file to an icon representing a trash can. Likewise, many word processors allow a user to move text in a document to another position by using only the mouse. We return to user interfaces in Chap. 3.

Many client-server applications can be constructed from roughly three different pieces: a part that handles interaction with a user, a part that operates on a database or file system, and a middle part that generally contains the core functionality of an application. This middle part is logically placed at the processing level. In contrast to user interfaces and databases, there are not many aspects common to the processing level. Therefore, we shall give several examples to make this level clearer.

As a first example, consider an Internet search engine. Ignoring all the animated banners, images, and other fancy window dressing, the user interface of a search engine is very simple: a user types in a string of keywords and is subsequently presented with a list of titles of Web pages. The back end is formed by a huge database of Web pages that have been prefetched and indexed. The core of the search engine is a program that transforms the user's string of keywords into one or more database queries. It subsequently ranks the results into a list, and transforms that list into a series of HTML pages. Within the client-server model, this information retrieval part is typically placed at the processing level. Fig. 2-4 shows this organization.

Figure 2-4. The simplified organization of an Internet search engine into three different layers.



As a second example, consider a decision support system for a stock brokerage. Analogous to a search engine, such a system can be divided into a front end implementing the user interface, a back end for accessing a database with the financial data, and the analysis programs between these two. Analysis of financial data may require sophisticated methods and techniques from statistics and artificial intelligence. In some cases, the core of a financial decision support system may even need to be executed on high-performance computers in order to achieve the throughput and responsiveness that is expected from its users.

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As a last example, consider a typical desktop package, consisting of a word processor, a spreadsheet application, communication facilities, and so on. Such "office" suites are generally integrated through a common user interface that supports compound documents, and operates on files from the user's home directory. (In an office environment, this home directory is often placed on a remote file server.) In this example, the processing level consists of a relatively large collection of programs, each having rather simple processing capabilities.

The data level in the client-server model contains the programs that maintain the actual data on which the applications operate. An important property of this level is that data are often persistent, that is, even if no application is running, data will be stored somewhere for next use. In its simplest form, the data level consists of a file system, but it is more common to use a full-fledged database. In the client-server model, the data level is typically implemented at the server side.

Besides merely storing data, the data level is generally also responsible for keeping data consistent across different applications. When databases are being used, maintaining consistency means that metadata such as table descriptions, entry constraints and application-

specific metadata are also stored at this level. For example, in the case of a bank, we may want to generate a notification when a customer's credit card debt reaches a certain value. This type of information can be maintained through a database trigger that activates a handler for that trigger at the appropriate moment.

In most business-oriented environments, the data level is organized as a relational database. Data independence is crucial here. The data are organized independent of the applications in such a way that changes in that organization do not affect applications, and neither do the applications affect the data organization. Using relational databases in the client-server model helps separate the processing level from the data level, as processing and data are considered independent.

However, relational databases are not always the ideal choice. A characteristic feature of many applications is that they operate on complex data types that are more easily modeled in terms of objects than in terms of relations. Examples of such data types range from simple polygons and circles to representations of aircraft designs, as is the case with computer-aided design (CAD) systems.

In those cases where data operations are more easily expressed in terms of object manipulations, it makes sense to implement the data level by means of an object-oriented or object-relational database. Notably the latter type has gained popularity as these databases build upon the widely dispersed relational data model, while offering the advantages that object-orientation gives.

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Multitiered Architectures

The distinction into three logical levels as discussed so far, suggests a number of possibilities for physically distributing a client-server application across several machines. The simplest organization is to have only two types of machines:

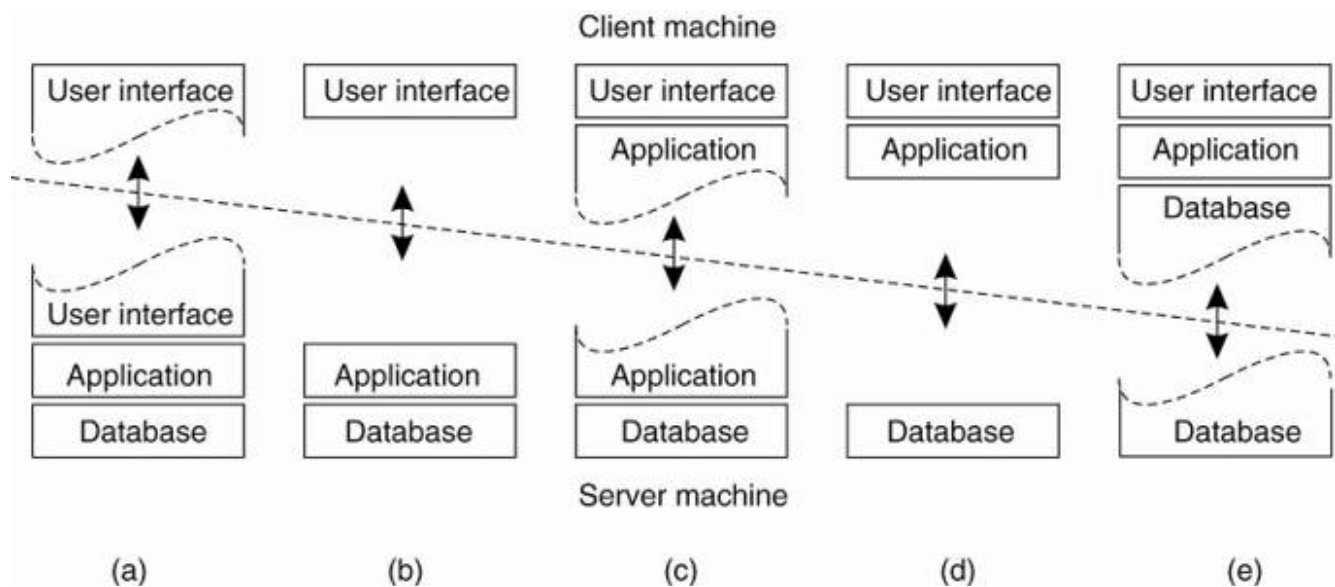
A client machine containing only the programs implementing (part of) the user-interface level

A server machine containing the rest, that is the programs implementing the processing and data level

In this organization everything is handled by the server while the client is essentially no more than a dumb terminal, possibly with a pretty graphical interface. There are many other possibilities, of which we explore some of the more common ones in this section.

One approach for organizing the clients and servers is to distribute the programs in the application layers of the previous section across different machines, as shown in Fig. 2-5 [see also Umar (1997); and Jing et al. (1999)]. As a first step, we make a distinction between only two kinds of machines: client machines and server machines, leading to what is also referred to as a (physically) two-tiered architecture.

Figure 2-5. Alternative client-server organizations (a)–(e).



One possible organization is to have only the terminal-dependent part of the user interface on the client machine, as shown in Fig. 2-5(a), and give the applications remote control over the presentation of their data. An alternative is to place the entire user-interface software on the client side, as shown in Fig. 2-5(b). In such cases, we essentially divide the application into a graphical front end, which communicates with the rest of the application (residing at the server) through an application-specific protocol. In this model, the front end (the client software) does no processing other than necessary for presenting the application's interface.

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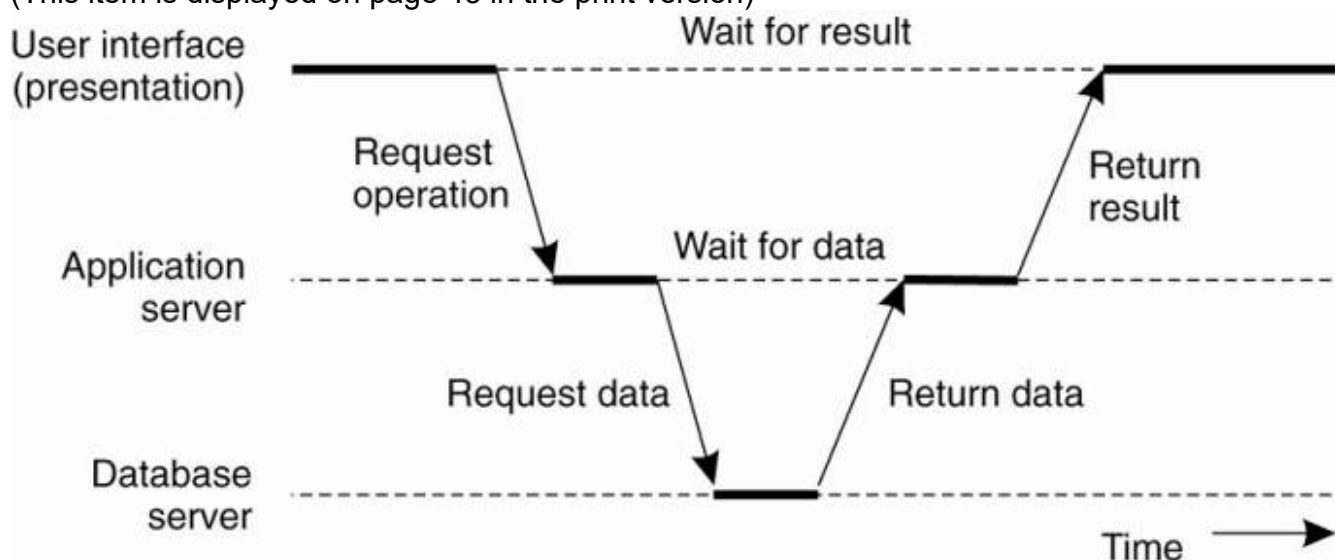
Continuing along this line of reasoning, we may also move part of the application to the front end, as shown in Fig. 2-5(c). An example where this makes sense is where the application makes use of a form that needs to be filled in entirely before it can be processed. The front end can then check the correctness and consistency of the form, and where necessary interact with the user. Another example of the organization of Fig. 2-5(c), is that of a word processor in which the basic editing functions execute on the client side where they operate on locally cached, or in-memory data, but where the advanced support tools such as checking the spelling and grammar execute on the server side.

In many client-server environments, the organizations shown in Fig. 2-5(d) and Fig. 2-5(e) are particularly popular. These organizations are used where the client machine is a PC or workstation, connected through a network to a distributed file system or database. Essentially, most of the application is running on the client machine, but all operations on files or database entries go to the server. For example, many banking applications run on an end-user's machine where the user prepares transactions and such. Once finished, the application contacts the database on the bank's server and uploads the transactions for further processing. Fig. 2-5(e) represents the situation where the client's local disk contains part of the data. For example, when browsing the Web, a client can gradually build a huge cache on local disk of most recent inspected Web pages.

We note that for a few years there has been a strong trend to move away from the configurations shown in Fig. 2-5(d) and Fig. 2-5(e) in those cases that client software is placed at end-user machines. In these cases, most of the processing and data storage is handled at the server side. The reason for this is simple: although client machines do a lot, they are also more problematic to manage. Having more functionality on the client machine makes client-side software more prone to errors and more dependent on the client's underlying platform (i.e., operating system and resources). From a system's management perspective, having what are called fat clients is not optimal. Instead the thin clients as represented by the organizations shown in Fig. 2-5(a)–(c) are much easier, perhaps at the cost of less sophisticated user interfaces and client-perceived performance.

Note that this trend does not imply that we no longer need distributed systems. On the contrary, what we are seeing is that server-side solutions are becoming increasingly more distributed as a single server is being replaced by multiple servers running on different machines. In particular, when distinguishing only client and server machines as we have done so far, we miss the point that a server may sometimes need to act as a client, as shown in Fig. 2-6, leading to a (physically) three-tiered architecture.

Figure 2-6. An example of a server acting as client.
(This item is displayed on page 43 in the print version)



In this architecture, programs that form part of the processing level reside on a separate server, but may additionally be partly distributed across the client and server machines. A typical example of where a three-tiered architecture is used is in transaction processing. As we discussed in Chap. 1, a separate process, called the transaction processing monitor, coordinates all transactions across possibly different data servers.

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Another, but very different example where we often see a three-tiered architecture is in the organization of Web sites. In this case, a Web server acts as an entry point to a site, passing requests to an application server where the actual processing takes place. This application server, in turn, interacts with a database server. For example, an application server may be

responsible for running the code to inspect the available inventory of some goods as offered by an electronic bookstore. To do so, it may need to interact with a database containing the raw inventory data. We will come back to Web site organization in Chap. 12.

2.2.2. Decentralized Architectures

Multitiered client-server architectures are a direct consequence of dividing applications into a user-interface, processing components, and a data level. The different tiers correspond directly with the logical organization of applications. In many business environments, distributed processing is equivalent to organizing a client-server application as a multitiered architecture. We refer to this type of distribution as vertical distribution. The characteristic feature of vertical distribution is that it is achieved by placing logically different components on different machines. The term is related to the concept of vertical fragmentation as used in distributed relational databases, where it means that tables are split column-wise, and subsequently distributed across multiple machines (Oszu and Valduriez, 1999).

Again, from a system management perspective, having a vertical distribution can help: functions are logically and physically split across multiple machines, where each machine is tailored to a specific group of functions. However, vertical distribution is only one way of organizing client-server applications. In modern architectures, it is often the distribution of the clients and the servers that counts, which we refer to as horizontal distribution. In this type of distribution, a client or server may be physically split up into logically equivalent parts, but each part is operating on its own share of the complete data set, thus balancing the load. In this section we will take a look at a class of modern system architectures that support horizontal distribution, known as peer-to-peer systems.

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From a high-level perspective, the processes that constitute a peer-to-peer system are all equal. This means that the functions that need to be carried out are represented by every process that constitutes the distributed system. As a consequence, much of the interaction between processes is symmetric: each process will act as a client and a server at the same time (which is also referred to as acting as a servent).

Given this symmetric behavior, peer-to-peer architectures evolve around the question how to organize the processes in an overlay network, that is, a network in which the nodes are formed by the processes and the links represent the possible communication channels (which are usually realized as TCP connections). In general, a process cannot communicate directly with an arbitrary other process, but is required to send messages through the available communication channels. Two types of overlay networks exist: those that are structured and those that are not. These two types are surveyed extensively in Lua et al. (2005) along with numerous examples. Aberer et al. (2005) provide a reference architecture that allows for a more formal comparison of the different types of peer-to-peer systems. A survey taken from the perspective of content distribution is provided by Androutsellis-Theotokis and Spinellis (2004).

Structured Peer-to-Peer Architectures

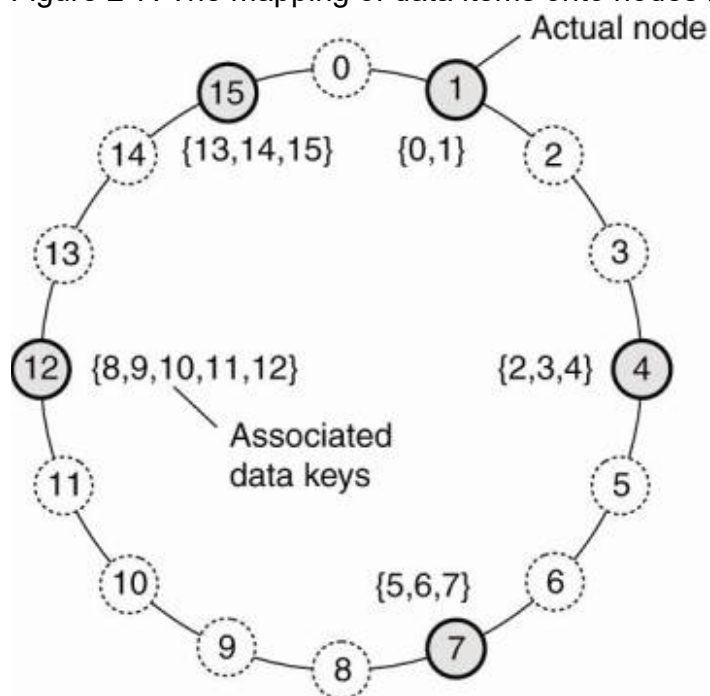
In a structured peer-to-peer architecture, the overlay network is constructed using a deterministic procedure. By far the most-used procedure is to organize the processes through a

distributed hash table (DHT). In a DHT-based system, data items are assigned a random key from a large identifier space, such as a 128-bit or 160-bit identifier. Likewise, nodes in the system are also assigned a random number from the same identifier space. The crux of every DHT-based system is then to implement an efficient and deterministic scheme that uniquely maps the key of a data item to the identifier of a node based on some distance metric (Balakrishnan, 2003). Most importantly, when looking up a data item, the network address of the node responsible for that data item is returned. Effectively, this is accomplished by routing a request for a data item to the responsible node.

For example, in the Chord system (Stoica et al., 2003) the nodes are logically organized in a ring such that a data item with key k is mapped to the node with the smallest identifier id_k . This node is referred to as the successor of key k and denoted as $succ(k)$, as shown in Fig. 2-7. To actually look up the data item, an application running on an arbitrary node would then call the function $LOOKUP(k)$ which would subsequently return the network address of $succ(k)$. At that point, the application can contact the node to obtain a copy of the data item.

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Figure 2-7. The mapping of data items onto nodes in Chord.



We will not go into algorithms for looking up a key now, but defer that discussion until Chap. 5 where we describe details of various naming systems. Instead, let us concentrate on how nodes organize themselves into an overlay network, or, in other words, membership management. In the following, it is important to realize that looking up a key does not follow the logical organization of nodes in the ring from Fig. 2-7. Rather, each node will maintain shortcuts to other nodes in such a way that lookups can generally be done in $O(\log(N))$ number of steps, where N is the number of nodes participating in the overlay.

Now consider Chord again. When a node wants to join the system, it starts with generating a random identifier id. Note that if the identifier space is large enough, then provided the random number generator is of good quality, the probability of generating an identifier that is already assigned to an actual node is close to zero. Then, the node can simply do a lookup on id, which will return the network address of succ(id). At that point, the joining node can simply contact succ(id) and its predecessor and insert itself in the ring. Of course, this scheme requires that each node also stores information on its predecessor. Insertion also yields that each data item whose key is now associated with node id, is transferred from succ(id).

Leaving is just as simple: node id informs its departure to its predecessor and successor, and transfers its data items to succ(id).

Similar approaches are followed in other DHT-based systems. As an example, consider the Content Addressable Network (CAN), described in Ratnasamy et al. (2001). CAN deploys a d-dimensional Cartesian coordinate space, which is completely partitioned among all the nodes that participate in the system. For purpose of illustration, let us consider only the 2-dimensional case, of which an example is shown in Fig. 2-8.

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Figure 2-8. (a) The mapping of data items onto nodes in CAN. (b) Splitting a region when a node joins.

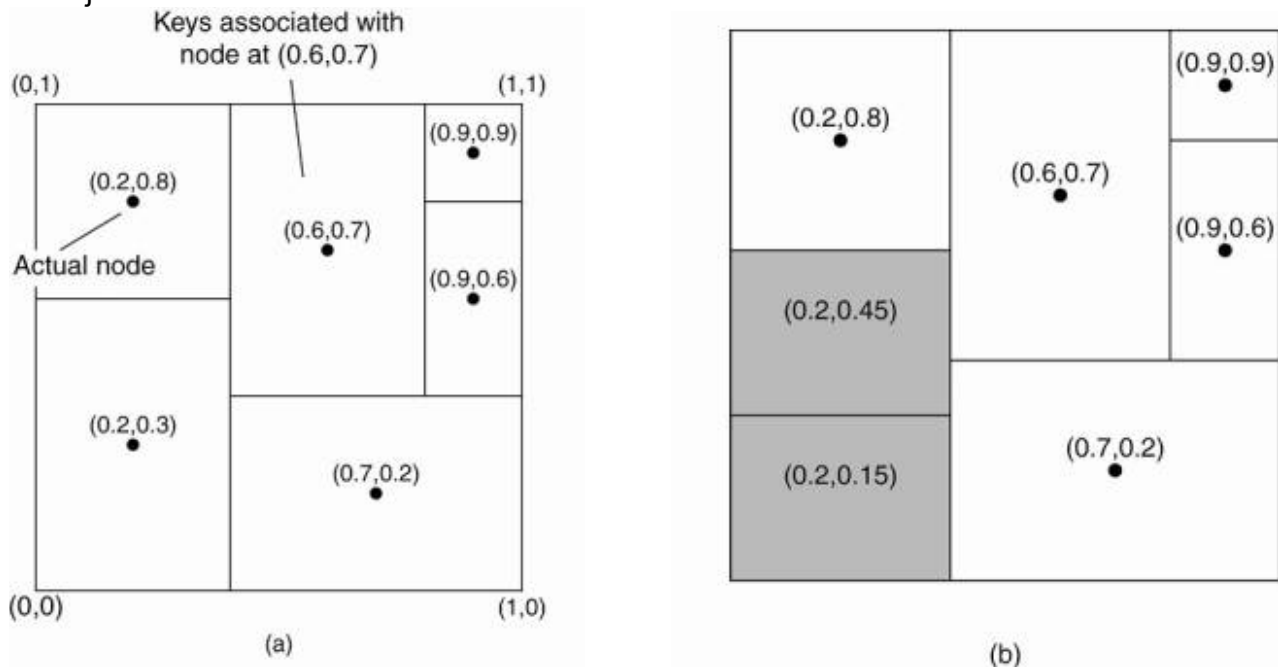


Fig. 2-8(a) shows how the two-dimensional space $[0,1] \times [0,1]$ is divided among six nodes. Each node has an associated region. Every data item in CAN will be assigned a unique point in this space, after which it is also clear which node is responsible for that data (ignoring data items that fall on the border of multiple regions, for which a deterministic assignment rule is used).

When a node P wants to join a CAN system, it picks an arbitrary point from the coordinate space and subsequently looks up the node Q in whose region that point falls. This lookup is

accomplished through positioned-based routing, of which the details are deferred until later chapters. Node Q then splits its region into two halves, as shown in Fig. 2-8(b), and one half is assigned to the node P. Nodes keep track of their neighbors, that is, nodes responsible for adjacent region. When splitting a region, the joining node P can easily come to know who its new neighbors are by asking node Q. As in Chord, the data items for which node P is now responsible are transferred from node Q.

Leaving is a bit more problematic in CAN. Assume that in Fig. 2-8, the node with coordinate (0.6,0.7) leaves. Its region will be assigned to one of its neighbors, say the node at (0.9,0.9), but it is clear that simply merging it and obtaining a rectangle cannot be done. In this case, the node at (0.9,0.9) will simply take care of that region and inform the old neighbors of this fact. Obviously, this may lead to less symmetric partitioning of the coordinate space, for which reason a background process is periodically started to repartition the entire space.

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Unstructured Peer-to-Peer Architectures

Unstructured peer-to-peer systems largely rely on randomized algorithms for constructing an overlay network. The main idea is that each node maintains a list of neighbors, but that this list is constructed in a more or less random way. Likewise, data items are assumed to be randomly placed on nodes. As a consequence, when a node needs to locate a specific data item, the only thing it can effectively do is flood the network with a search query (Rismon and Moors, 2006). We will return to searching in unstructured overlay networks in Chap. 5, and for now concentrate on membership management.

One of the goals of many unstructured peer-to-peer systems is to construct an overlay network that resembles a random graph. The basic model is that each node maintains a list of c neighbors, where, ideally, each of these neighbors represents a randomly chosen live node from the current set of nodes. The list of neighbors is also referred to as a partial view. There are many ways to construct such a partial view. Jelasi et al. (2004, 2005a) have developed a framework that captures many different algorithms for overlay construction to allow for evaluations and comparison. In this framework, it is assumed that nodes regularly exchange entries from their partial view. Each entry identifies another node in the network, and has an associated age that indicates how old the reference to that node is. Two threads are used, as shown in Fig. 2-9.

The active thread takes the initiative to communicate with another node. It selects that node from its current partial view. Assuming that entries need to be pushed to the selected peer, it continues by constructing a buffer containing $c/2+1$ entries, including an entry identifying itself. The other entries are taken from the current partial view.

If the node is also in pull mode it will wait for a response from the selected peer. That peer, in the meantime, will also have constructed a buffer by means the passive thread shown in Fig. 2-9(b), whose activities strongly resemble that of the active thread.

The crucial point is the construction of a new partial view. This view, for initiating as well as for the contacted peer, will contain exactly c entries, part of which will come from received buffer. In essence, there are two ways to construct the new view. First, the two nodes may decide to discard the entries that they had sent to each other. Effectively, this means that they will swap

part of their original views. The second approach is to discard as many old entries as possible. In general, it turns out that the two approaches are complementary [see Jelasity et al. (2005a) for the details]. It turns out that many membership management protocols for unstructured overlays fit this framework. There are a number of interesting observations to make.

First, let us assume that when a node wants to join it contacts an arbitrary other node, possibly from a list of well-known access points. This access point is just a regular member of the overlay, except that we can assume it to be highly available. In this case, it turns out that protocols that use only push mode or only pull mode can fairly easily lead to disconnected overlays. In other words, groups of nodes will become isolated and will never be able to reach every other node in the network. Clearly, this is an undesirable feature, for which reason it makes more sense to let nodes actually exchange entries.

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Figure 2-9. (a) The steps taken by the active thread. (b) The steps take by the passive thread.

Actions by active thread (periodically repeated):

```
select a peer P from the current partial view;
if PUSH_MODE {
    mybuffer = [(MyAddress, 0)];
    permute partial view;
    move H oldest entries to the end;
    append first c/2 entries to mybuffer;
    send mybuffer to P;
} else {
    send trigger to P;
}
if PULL_MODE {
    receive P's buffer;
}
construct a new partial view from the current one and P's
buffer;
increment the age of every entry in the new partial view;
```

(a)

Actions by passive thread:

```
receive buffer from any process Q;
if PULL_MODE {
    mybuffer = [(MyAddress, 0)];
    permute partial vie w;
    move H oldest entries to the end;
    append first c/2 entries to mybuffer;
    send mybuffer to P;
}
construct a new partial view from the current one and P's
buffer;
increment the age of every entry in the new partial view;
```


(b)

Second, leaving the network turns out to be a very simple operation provided the nodes exchange partial views on a regular basis. In this case, a node can simply depart without informing any other node. What will happen is that when a node P selects one of its apparent neighbors, say node Q, and discovers that Q no longer responds, it simply removes the entry from its partial view to select another peer. It turns out that when constructing a new partial view, a node follows the policy to discard as many old entries as possible, departed nodes will rapidly be forgotten. In other words, entries referring to departed nodes will automatically be quickly removed from partial views.

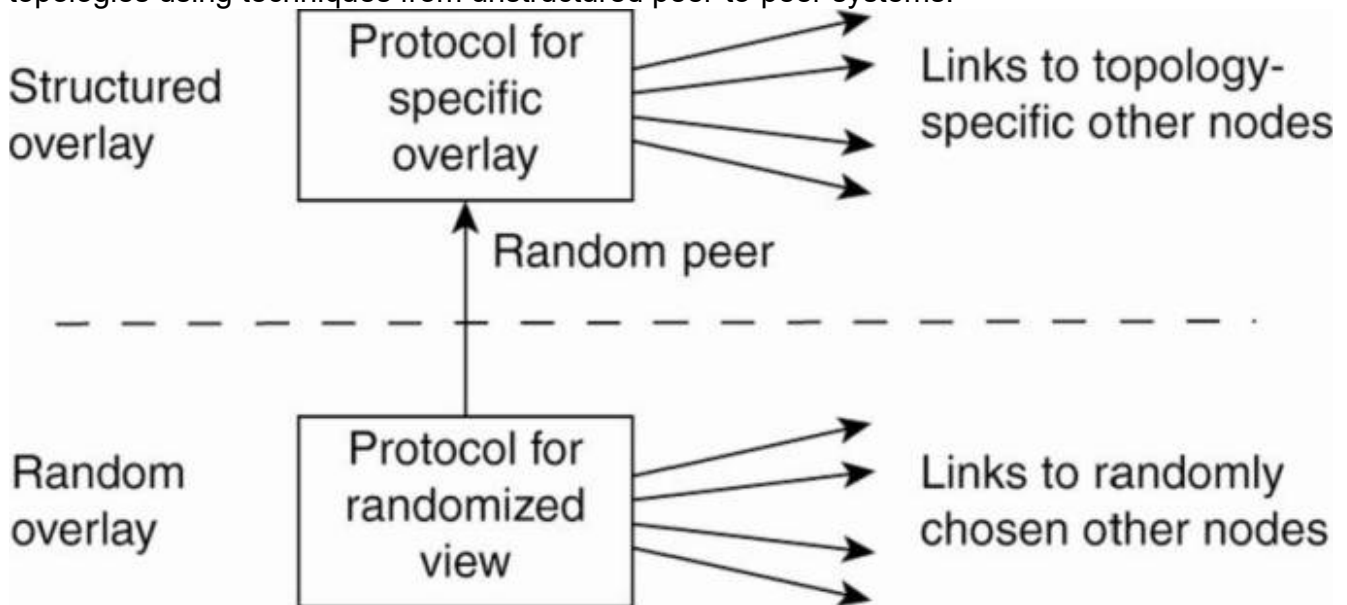
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However, there is a price to pay when this strategy is followed. To explain, consider for a node P the set of nodes that have an entry in their partial view that refers to P. Technically, this is known as the indegree of a node. The higher node P's indegree is, the higher the probability that some other node will decide to contact P. In other words, there is a danger that P will become a popular node, which could easily bring it into an imbalanced position regarding workload. Systematically discarding old entries turns out to promote nodes to ones having a high indegree. There are other trade-offs in addition, for which we refer to Jelasity et al. (2005a).

Topology Management of Overlay Networks

Although it would seem that structured and unstructured peer-to-peer systems form strict independent classes, this need actually not be case [see also Castro et al. (2005)]. One key observation is that by carefully exchanging and selecting entries from partial views, it is possible to construct and maintain specific topologies of overlay networks. This topology management is achieved by adopting a two-layered approach, as shown in Fig. 2-10.

Figure 2-10. A two-layered approach for constructing and maintaining specific overlay topologies using techniques from unstructured peer-to-peer systems.



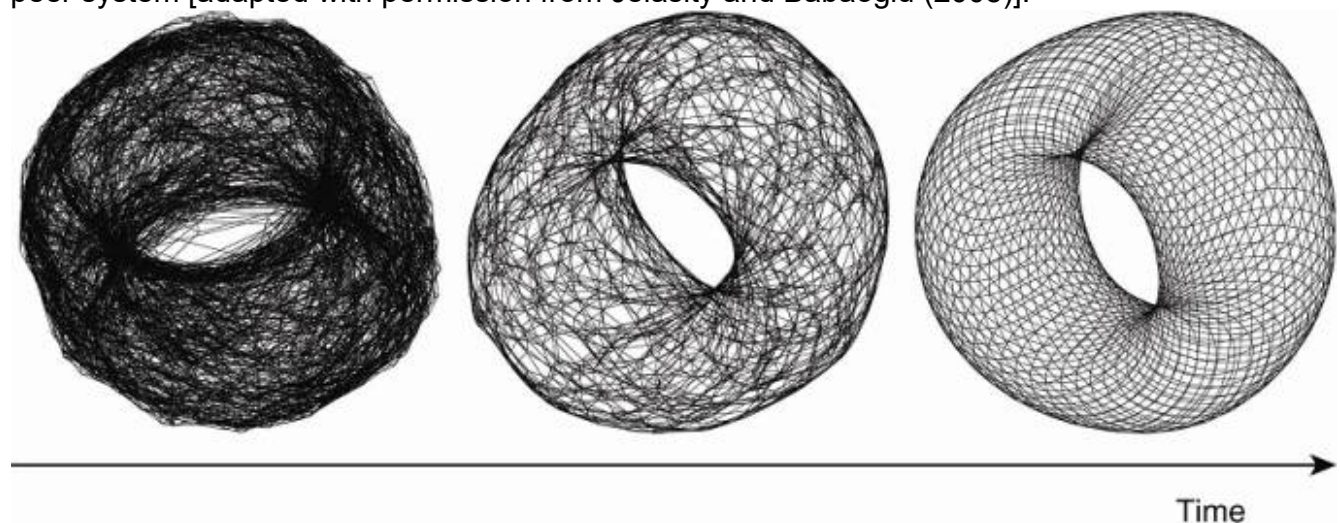
The lowest layer constitutes an unstructured peer-to-peer system in which nodes periodically exchange entries of their partial views with the aim to maintain an accurate random graph. Accuracy in this case refers to the fact that the partial view should be filled with entries referring to randomly selected live nodes.

The lowest layer passes its partial view to the higher layer, where an additional selection of entries takes place. This then leads to a second list of neighbors corresponding to the desired topology. Jelasity and Babaoglu (2005) propose to use a ranking function by which nodes are ordered according to some criterion relative to a given node. A simple ranking function is to order a set of nodes by increasing distance from a given node P . In that case, node P will gradually build up a list of its nearest neighbors, provided the lowest layer continues to pass randomly selected nodes.

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As an illustration, consider a logical grid of size $N \times N$ with a node placed on each point of the grid. Every node is required to maintain a list of c nearest neighbors, where the distance between a node at (a_1, a_2) and (b_1, b_2) is defined as $d_1 + d_2$, with $d_i = \min(N - |a_i - b_i|, |a_i - b_i|)$. If the lowest layer periodically executes the protocol as outlined in Fig. 2-9, the topology that will evolve is a torus, shown in Fig. 2-11.

Figure 2-11. Generating a specific overlay network using a two-layered unstructured peer-to-peer system [adapted with permission from Jelasity and Babaoglu (2005)].



Of course, completely different ranking functions can be used. Notably those that are related to capturing the semantic proximity of the data items as stored at a peer node are interesting. This proximity allows for the construction of semantic overlay networks that allow for highly efficient search algorithms in unstructured peer-to-peer systems. We will return to these systems in Chap. 5 when we discuss attribute-based naming.

Superpeers

Notably in unstructured peer-to-peer systems, locating relevant data items can become problematic as the network grows. The reason for this scalability problem is simple: as there is

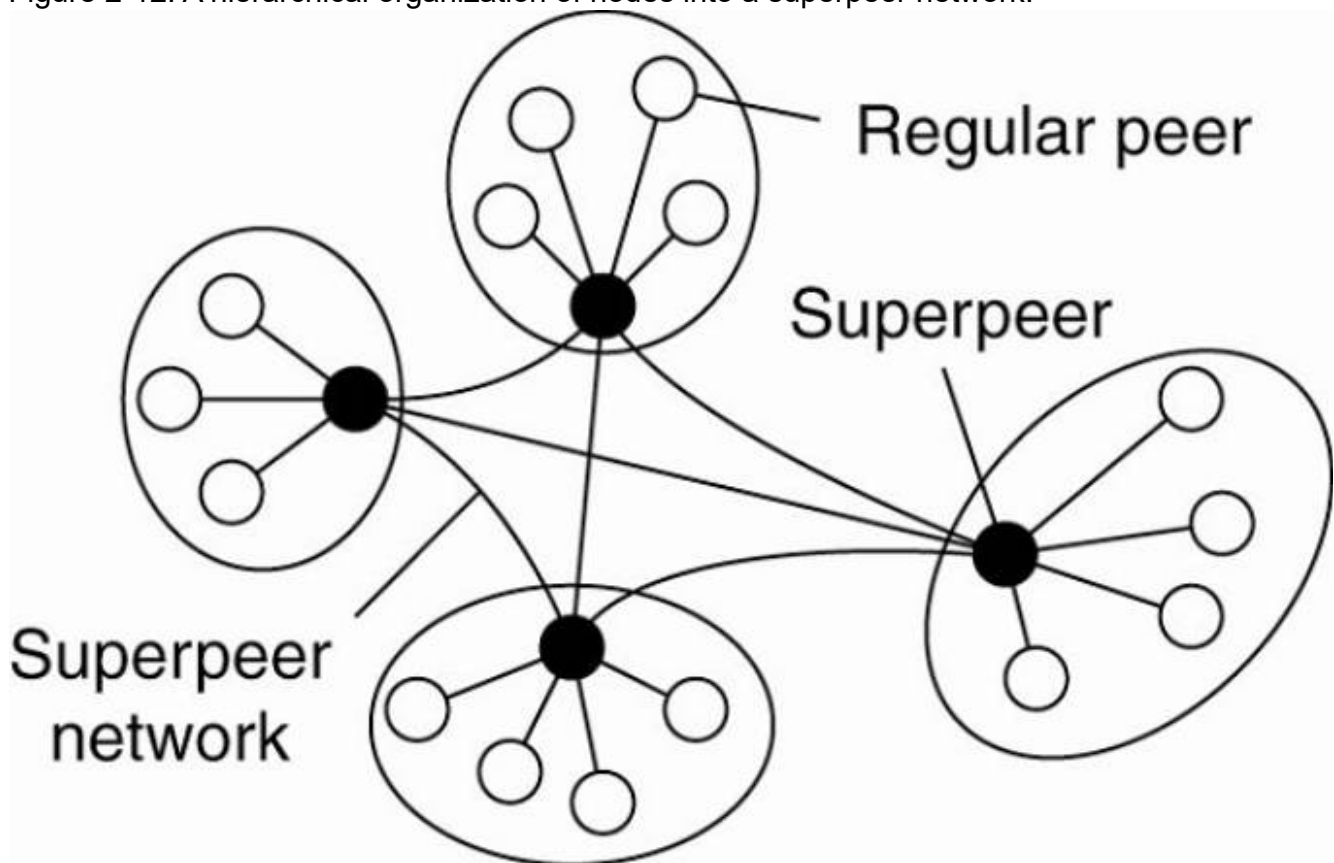
no deterministic way of routing a lookup request to a specific data item, essentially the only technique a node can resort to is flooding the request. There are various ways in which flooding can be dammed, as we will discuss in Chap. 5, but as an alternative many peer-to-peer systems have proposed to make use of special nodes that maintain an index of data items.

There are other situations in which abandoning the symmetric nature of peer-to-peer systems is sensible. Consider a collaboration of nodes that offer resources to each other. For example, in a collaborative content delivery network (CDN), nodes may offer storage for hosting copies of Web pages allowing Web clients to access pages nearby, and thus to access them quickly. In this case a node P may need to seek for resources in a specific part of the network. In that case, making use of a broker that collects resource usage for a number of nodes that are in each other's proximity will allow to quickly select a node with sufficient resources.

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Nodes such as those maintaining an index or acting as a broker are generally referred to as superpeers. As their name suggests, superpeers are often also organized in a peer-to-peer network, leading to a hierarchical organization as explained in Yang and Garcia-Molina (2003). A simple example of such an organization is shown in Fig. 2-12. In this organization, every regular peer is connected as a client to a superpeer. All communication from and to a regular peer proceeds through that peer's associated superpeer.

Figure 2-12. A hierarchical organization of nodes into a superpeer network.



In many cases, the client-superpeer relation is fixed: whenever a regular peer joins the network, it attaches to one of the superpeers and remains attached until it leaves the network. Obviously, it is expected that superpeers are long-lived processes with a high availability. To compensate for potential unstable behavior of a superpeer, backup schemes can be deployed, such as pairing every superpeer with another one and requiring clients to attach to both.

Having a fixed association with a superpeer may not always be the best solution. For example, in the case of file-sharing networks, it may be better for a client to attach to a superpeer that maintains an index of files that the client is generally interested in. In that case, chances are bigger that when a client is looking for a specific file, its superpeer will know where to find it. Garbacki et al. (2005) describe a relatively simple scheme in which the client-superpeer relation can change as clients discover better superpeers to associate with. In particular, a superpeer returning the result of a lookup operation is given preference over other superpeers.

As we have seen, peer-to-peer networks offer a flexible means for nodes to join and leave the network. However, with superpeer networks a new problem is introduced, namely how to select the nodes that are eligible to become superpeer. This problem is closely related to the leader-election problem, which we discuss in Chap. 6, when we return to electing superpeers in a peer-to-peer network.

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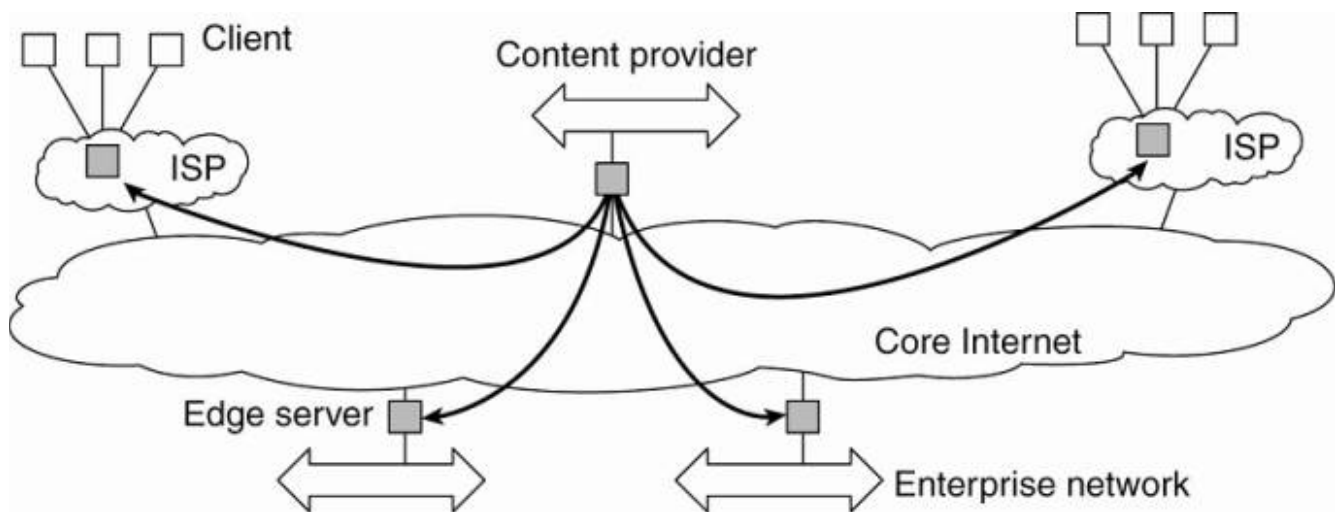
2.2.3. Hybrid Architectures

So far, we have focused on client-server architectures and a number of peer-to-peer architectures. Many distributed systems combine architectural features, as we already came across in superpeer networks. In this section we take a look at some specific classes of distributed systems in which client-server solutions are combined with decentralized architectures.

Edge-Server Systems

An important class of distributed systems that is organized according to a hybrid architecture is formed by edge-server systems. These systems are deployed on the Internet where servers are placed "at the edge" of the network. This edge is formed by the boundary between enterprise networks and the actual Internet, for example, as provided by an Internet Service Provider (ISP). Likewise, where end users at home connect to the Internet through their ISP, the ISP can be considered as residing at the edge of the Internet. This leads to a general organization as shown in Fig. 2-13.

Figure 2-13. Viewing the Internet as consisting of a collection of edge servers.



End users, or clients in general, connect to the Internet by means of an edge server. The edge server's main purpose is to serve content, possibly after applying filtering and transcoding functions. More interesting is the fact that a collection of edge servers can be used to optimize content and application distribution. The basic model is that for a specific organization, one edge server acts as an origin server from which all content originates. That server can use other edge servers for replicating Web pages and such (Leff et al., 2004; Nayate et al., 2004; and Rabinovich and Spatscheck, 2002). We will return to edge-server systems in Chap. 12 when we discuss Web-based solutions.

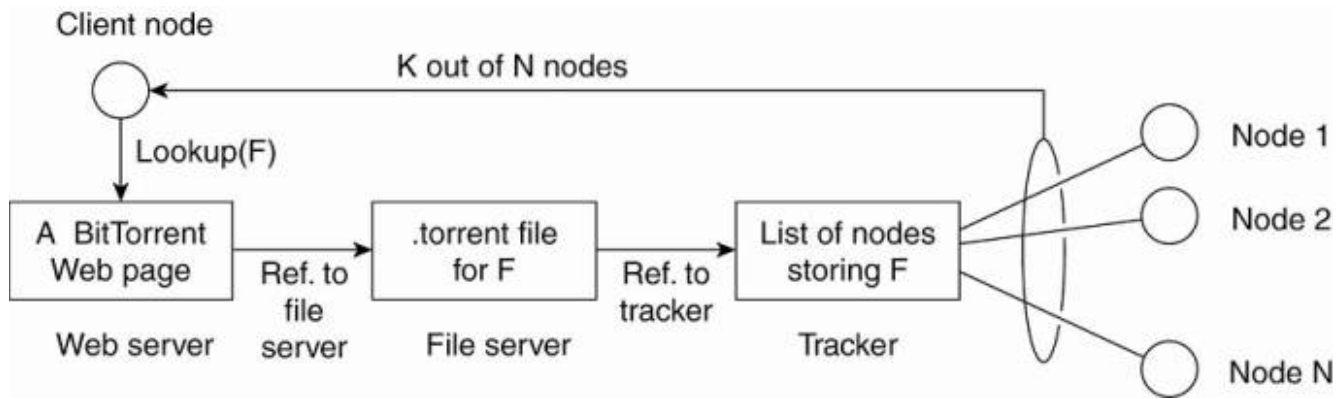
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Collaborative Distributed Systems

Hybrid structures are notably deployed in collaborative distributed systems. The main issue in many of these systems is to first get started, for which often a traditional client-server scheme is deployed. Once a node has joined the system, it can use a fully decentralized scheme for collaboration.

To make matters concrete, let us first consider the BitTorrent file-sharing system (Cohen, 2003). BitTorrent is a peer-to-peer file downloading system. Its principal working is shown in Fig. 2-14. The basic idea is that when an end user is looking for a file, he downloads chunks of the file from other users until the downloaded chunks can be assembled together yielding the complete file. An important design goal was to ensure collaboration. In most file-sharing systems, a significant fraction of participants merely download files but otherwise contribute close to nothing (Adar and Huberman, 2000; Saroiu et al., 2003; and Yang et al., 2005). To this end, a file can be downloaded only when the downloading client is providing content to someone else. We will return to this "tit-for-tat" behavior shortly.

Figure 2-14. The principal working of BitTorrent [adapted with permission from Pouwelse et al. (2004)].



To download a file, a user needs to access a global directory, which is just one of a few well-known Web sites. Such a directory contains references to what are called .torrent files. A .torrent file contains the information that is needed to download a specific file. In particular, it refers to what is known as a tracker, which is a server that is keeping an accurate account of active nodes that have (chunks) of the requested file. An active node is one that is currently downloading another file. Obviously, there will be many different trackers, although there will generally be only a single tracker per file (or collection of files).

Once the nodes have been identified from where chunks can be downloaded, the downloading node effectively becomes active. At that point, it will be forced to help others, for example by providing chunks of the file it is downloading that others do not yet have. This enforcement comes from a very simple rule: if node P notices that node Q is downloading more than it is uploading, P can decide to decrease the rate at which it sends data to Q. This scheme works well provided P has something to download from Q. For this reason, nodes are often supplied with references to many other nodes putting them in a better position to trade data.

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Clearly, BitTorrent combines centralized with decentralized solutions. As it turns out, the bottleneck of the system is, not surprisingly, formed by the trackers.

As another example, consider the Globule collaborative content distribution network (Pierre and van Steen, 2006). Globule strongly resembles the edge-server architecture mentioned above. In this case, instead of edge servers, end users (but also organizations) voluntarily provide enhanced Web servers that are capable of collaborating in the replication of Web pages. In its simplest form, each such server has the following components:

1. A component that can redirect client requests to other servers.
2. A component for analyzing access patterns.
3. A component for managing the replication of Web pages.

The server provided by Alice is the Web server that normally handles the traffic for Alice's Web site and is called the origin server for that site. It collaborates with other servers, for example, the one provided by Bob, to host the pages from Bob's site. In this sense, Globule is a decentralized distributed system. Requests for Alice's Web site are initially forwarded to her

server, at which point they may be redirected to one of the other servers. Distributed redirection is also supported.

However, Globule also has a centralized component in the form of its broker. The broker is responsible for registering servers, and making these servers known to others. Servers communicate with the broker completely analogous to what one would expect in a client-server system. For reasons of availability, the broker can be replicated, but as we shall later in this book, this type of replication is widely applied in order to achieve reliable client-server computing.

2.3. Architectures Versus Middleware

When considering the architectural issues we have discussed so far, a question that comes to mind is where middleware fits in. As we discussed in Chap. 1, middleware forms a layer between applications and distributed platforms, as shown in Fig. 1-1. An important purpose is to provide a degree of distribution transparency, that is, to a certain extent hiding the distribution of data, processing, and control from applications.

What is commonly seen in practice is that middleware systems actually follow a specific architectural style. For example, many middleware solutions have adopted an object-based architectural style, such as CORBA (OMG, 2004a). Others, like TIB/Rendezvous (TIBCO, 2005) provide middleware that follows the event-based architectural style. In later chapters, we will come across more examples of architectural styles.

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Having middleware molded according to a specific architectural style has the benefit that designing applications may become simpler. However, an obvious drawback is that the middleware may no longer be optimal for what an application developer had in mind. For example, CORBA initially offered only objects that could be invoked by remote clients. Later, it was felt that having only this form of interaction was too restrictive, so that other interaction patterns such as messaging were added. Obviously, adding new features can easily lead to bloated middle-ware solutions.

In addition, although middleware is meant to provide distribution transparency, it is generally felt that specific solutions should be adaptable to application requirements. One solution to this problem is to make several versions of a middleware system, where each version is tailored to a specific class of applications. An approach that is generally considered better is to make middleware systems such that they are easy to configure, adapt, and customize as needed by an application. As a result, systems are now being developed in which a stricter separation between policies and mechanisms is being made. This has led to several mechanisms by which the behavior of middleware can be modified (Sadjadi and McKinley, 2003). Let us take a look at some of the commonly followed approaches.

2.3.1. Interceptors

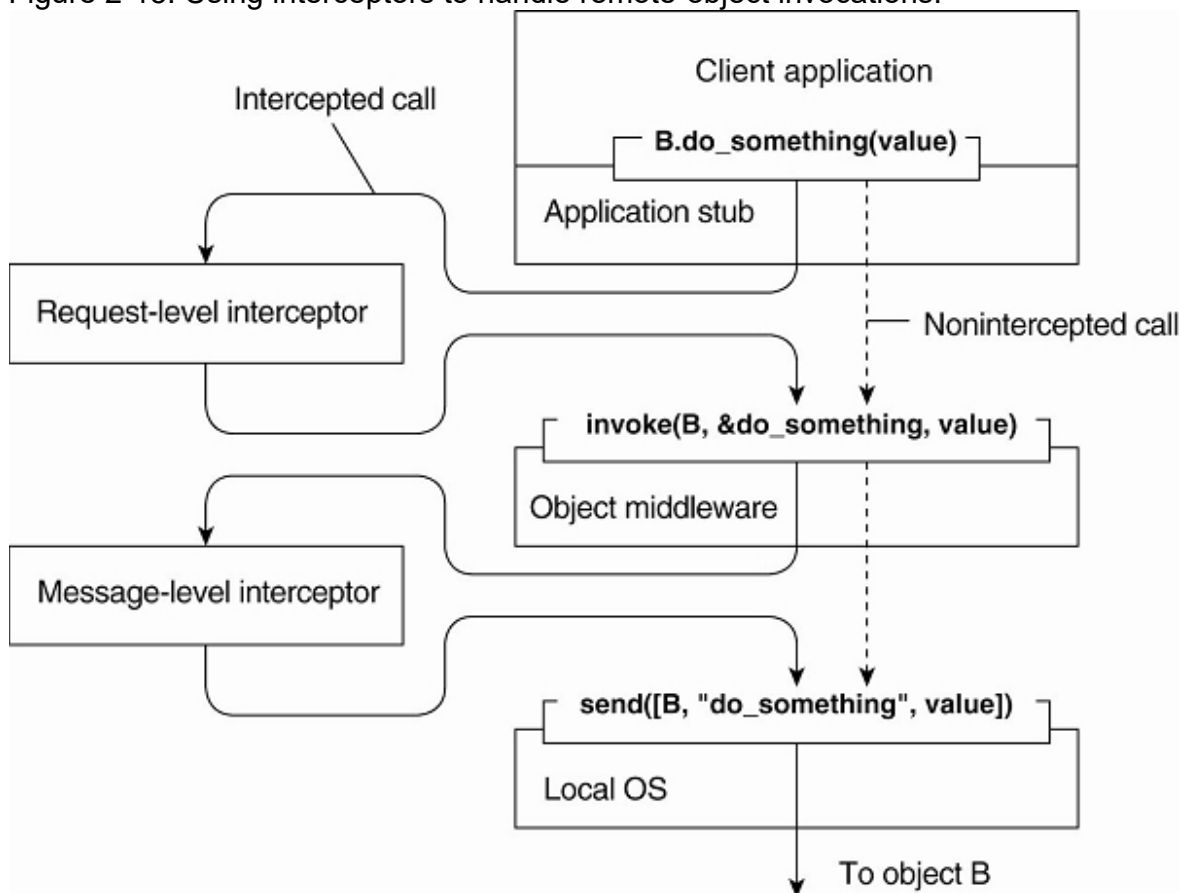
Conceptually, an interceptor is nothing but a software construct that will break the usual flow of control and allow other (application specific) code to be executed. To make interceptors generic may require a substantial implementation effort, as illustrated in Schmidt et al. (2000), and it is unclear whether in such cases generality should be preferred over restricted applicability and

simplicity. Also, in many cases having only limited interception facilities will improve management of the software and the distributed system as a whole.

To make matters concrete, consider interception as supported in many object-based distributed systems. The basic idea is simple: an object A can call a method that belongs to an object B, while the latter resides on a different machine than A. As we explain in detail later in the book, such a remote-object invocation is carried as a three-step approach:

1. Object A is offered a local interface that is exactly the same as the interface offered by object B. A simply calls the method available in that interface.
2. The call by A is transformed into a generic object invocation, made possible through a general object-invocation interface offered by the middleware at the machine where A resides.
3. Finally, the generic object invocation is transformed into a message that is sent through the transport-level network interface as offered by A's local operating system.

Figure 2-15. Using interceptors to handle remote-object invocations.



After the first step, the call `B.do_something(value)` is transformed into a generic call such as `invoke(B, &do_something, value)` with a reference to B's method and the parameters that go along with the call. Now imagine that object B is replicated. In that case, each replica should actually be invoked. This is a clear point where interception can help. What the request-level

interceptor will do is simply call `invoke(B, &do_something, value)` for each of the replicas. The beauty of this all is that the object A need not be aware of the replication of B, but also the object middleware need not have special components that deal with this replicated call. Only the request-level interceptor, which may be added to the middleware needs to know about B's replication.

In the end, a call to a remote object will have to be sent over the network. In practice, this means that the messaging interface as offered by the local operating system will need to be invoked. At that level, a message-level interceptor may assist in transferring the invocation to the target object. For example, imagine that the parameter value actually corresponds to a huge array of data. In that case, it may be wise to fragment the data into smaller parts to have it assembled again at the destination. Such a fragmentation may improve performance or reliability. Again, the middleware need not be aware of this fragmentation; the lower-level interceptor will transparently handle the rest of the communication with the local operating system.

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2.3.2. General Approaches to Adaptive Software

What interceptors actually offer is a means to adapt the middleware. The need for adaptation comes from the fact that the environment in which distributed applications are executed changes continuously. Changes include those resulting from mobility, a strong variance in the quality-of-service of networks, failing hardware, and battery drainage, amongst others. Rather than making applications responsible for reacting to changes, this task is placed in the middleware.

These strong influences from the environment have brought many designers of middleware to consider the construction of adaptive software. However, adaptive software has not been as successful as anticipated. As many researchers and developers consider it to be an important aspect of modern distributed systems, let us briefly pay some attention to it. McKinley et al. (2004) distinguish three basic techniques to come to software adaptation:

1. Separation of concerns
2. Computational reflection
3. Component-based design

Separating concerns relates to the traditional way of modularizing systems: separate the parts that implement functionality from those that take care of other things (known as extra functionalities) such as reliability, performance, security, etc. One can argue that developing middleware for distributed applications is largely about handling extra functionalities independent from applications. The main problem is that we cannot easily separate these extra functionalities by means of modularization. For example, simply putting security into a separate module is not going to work. Likewise, it is hard to imagine how fault tolerance can be isolated into a separate box and sold as an independent service. Separating and subsequently weaving these cross-cutting concerns into a (distributed) system is the major theme addressed by aspect-oriented software development (Filman et al., 2005). However, aspect orientation has

not yet been successfully applied to developing large-scale distributed systems, and it can be expected that there is still a long way to go before it reaches that stage.

Computational reflection refers to the ability of a program to inspect itself and, if necessary, adapt its behavior (Kon et al., 2002). Reflection has been built into programming languages, including Java, and offers a powerful facility for runtime modifications. In addition, some middleware systems provide the means to apply reflective techniques. However, just as in the case of aspect orientation, reflective middleware has yet to prove itself as a powerful tool to manage the complexity of large-scale distributed systems. As mentioned by Blair et al. (2004), applying reflection to a broad domain of applications is yet to be done.

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Finally, component-based design supports adaptation through composition. A system may either be configured statically at design time, or dynamically at runtime. The latter requires support for late binding, a technique that has been successfully applied in programming language environments, but also for operating systems where modules can be loaded and unloaded at will. Research is now well underway to allow automatic selection of the best implementation of a component during runtime (Yellin, 2003), but again, the process remains complex for distributed systems, especially when considering that replacement of one component requires knowing what the effect of that replacement on other components will be. In many cases, components are less independent as one may think.

2.3.3. Discussion

Software architectures for distributed systems, notably found as middleware, are bulky and complex. In large part, this bulkiness and complexity arises from the need to be general in the sense that distribution transparency needs to be provided. At the same time applications have specific extra-functional requirements that conflict with aiming at fully achieving this transparency. These conflicting requirements for generality and specialization have resulted in middleware solutions that are highly flexible. The price to pay, however, is complexity. For example, Zhang and Jacobsen (2004) report a 50% increase in the size of a particular software product in just four years since its introduction, whereas the total number of files for that product had tripled during the same period. Obviously, this is not an encouraging direction to pursue.

Considering that virtually all large software systems are nowadays required to execute in a networked environment, we can ask ourselves whether the complexity of distributed systems is simply an inherent feature of attempting to make distribution transparent. Of course, issues such as openness are equally important, but the need for flexibility has never been so prevalent as in the case of middleware.

Coyler et al. (2003) argue that what is needed is a stronger focus on (external) simplicity, a simpler way to construct middleware by components, and application independence. Whether any of the techniques mentioned above forms the solution is subject to debate. In particular, none of the proposed techniques so far have found massive adoption, nor have they been successfully applied to large-scale systems.

The underlying assumption is that we need adaptive software in the sense that the software should be allowed to change as the environment changes. However, one should question whether adapting to a changing environment is a good reason to adopt changing the software.

Faulty hardware, security attacks, energy drainage, and so on, all seem to be environmental influences that can (and should) be anticipated by software.

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The strongest, and certainly most valid, argument for supporting adaptive software is that many distributed systems cannot be shut down. This constraint calls for solutions to replace and upgrade components on the fly, but is not clear whether any of the solutions proposed above are the best ones to tackle this maintenance problem.

What then remains is that distributed systems should be able to react to changes in their environment by, for example, switching policies for allocating resources. All the software components to enable such an adaptation will already be in place. It is the algorithms contained in these components and which dictate the behavior that change their settings. The challenge is to let such reactive behavior take place without human intervention. This approach is seen to work better when discussing the physical organization of distributed systems when decisions are taken about where components are placed, for example. We discuss such system architectural issues next.

2.4. Self-Management in Distributed Systems

Distributed systems—and notably their associated middleware—need to provide general solutions toward shielding undesirable features inherent to networking so that they can support as many applications as possible. On the other hand, full distribution transparency is not what most applications actually want, resulting in application-specific solutions that need to be supported as well. We have argued that, for this reason, distributed systems should be adaptive, but notably when it comes to adapting their execution behavior and not the software components they comprise.

When adaptation needs to be done automatically, we see a strong interplay between system architectures and software architectures. On the one hand, we need to organize the components of a distributed system such that monitoring and adjustments can be done, while on the other hand we need to decide where the processes are to be executed that handle the adaptation.

In this section we pay explicit attention to organizing distributed systems as high-level feedback-control systems allowing automatic adaptations to changes. This phenomenon is also known as autonomic computing (Kephart, 2003) or self-star systems (Babaoglu et al., 2005). The latter name indicates the variety by which automatic adaptations are being captured: self-managing, self-healing, self-configuring, self-optimizing, and so on. We resort simply to using the name self-managing systems as coverage of its many variants.

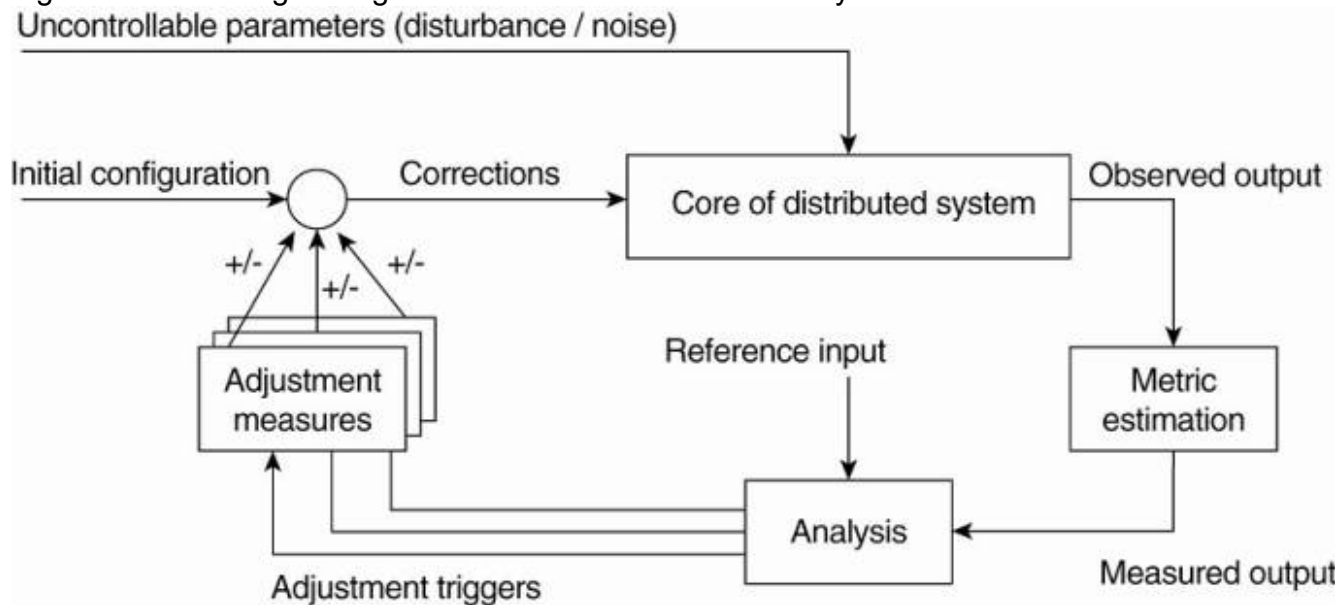
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2.4.1. The Feedback Control Model

There are many different views on self-managing systems, but what most have in common (either explicitly or implicitly) is the assumption that adaptations take place by means of one or more feedback control loops. Accordingly, systems that are organized by means of such loops are referred to as feedback control systems. Feedback control has since long been applied in various engineering fields, and its mathematical foundations are gradually also finding their way

in computing systems (Hellerstein et al., 2004; and Diao et al., 2005). For self-managing systems, the architectural issues are initially the most interesting. The basic idea behind this organization is quite simple, as shown in Fig. 2-16.

Figure 2-16. The logical organization of a feedback control system.



The core of a feedback control system is formed by the components that need to be managed. These components are assumed to be driven through controllable input parameters, but their behavior may be influenced by all kinds of uncontrollable input, also known as disturbance or noise input. Although disturbance will often come from the environment in which a distributed system is executing, it may well be the case that unanticipated component interaction causes unexpected behavior.

There are essentially three elements that form the feedback control loop. First, the system itself needs to be monitored, which requires that various aspects of the system need to be measured. In many cases, measuring behavior is easier said than done. For example, round-trip delays in the Internet may vary wildly, and also depend on what exactly is being measured. In such cases, accurately estimating a delay may be difficult indeed. Matters are further complicated when a node A needs to estimate the latency between two other completely different nodes B and C, without being able to intrude on either two nodes. For reasons as this, a feedback control loop generally contains a logical metric estimation component.

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Another part of the feedback control loop analyzes the measurements and compares these to reference values. This feedback analysis component forms the heart of the control loop, as it will contain the algorithms that decide on possible adaptations.

The last group of components consist of various mechanisms to directly influence the behavior of the system. There can be many different mechanisms: placing replicas, changing scheduling priorities, switching services, moving data for reasons of availability, redirecting requests to different servers, etc. The analysis component will need to be aware of these mechanisms and

their (expected) effect on system behavior. Therefore, it will trigger one or several mechanisms, to subsequently later observe the effect.

An interesting observation is that the feedback control loop also fits the manual management of systems. The main difference is that the analysis component is replaced by human administrators. However, in order to properly manage any distributed system, these administrators will need decent monitoring equipment as well as decent mechanisms to control the behavior of the system. It should be clear that properly analyzing measured data and triggering the correct actions makes the development of self-managing systems so difficult.

It should be stressed that Fig. 2-16 shows the logical organization of a self-managing system, and as such corresponds to what we have seen when discussing software architectures. However, the physical organization may be very different. For example, the analysis component may be fully distributed across the system. Likewise, taking performance measurements are usually done at each machine that is part of the distributed system. Let us now take a look at a few concrete examples on how to monitor, analyze, and correct distributed systems in an automatic fashion. These examples will also illustrate this distinction between logical and physical organization.

2.4.2. Example: Systems Monitoring with Astrolabe

As our first example, we consider Astrolabe (Van Renesse et al., 2003), which is a system that can support general monitoring of very large distributed systems. In the context of self-managing systems, Astrolabe is to be positioned as a general tool for observing systems behavior. Its output can be used to feed into an analysis component for deciding on corrective actions.

Astrolabe organizes a large collection of hosts into a hierarchy of zones. The lowest-level zones consist of just a single host, which are subsequently grouped into zones of increasing size. The top-level zone covers all hosts. Every host runs an Astrolabe process, called an agent, that collects information on the zones in which that host is contained. The agent also communicates with other agents with the aim to spread zone information across the entire system.

Each host maintains a set of attributes for collecting local information. For example, a host may keep track of specific files it stores, its resource usage, and so on. Only the attributes as maintained directly by hosts, that is, at the lowest level of the hierarchy are writable. Each zone can also have a collection of attributes, but the values of these attributes are computed from the values of lower level zones.

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Consider the following simple example shown in Fig. 2-17 with three hosts, A, B, and C grouped into a zone. Each machine keeps track of its IP address, CPU load, available free memory, and the number of active processes. Each of these attributes can be directly written using local information from each host. At the zone level, only aggregated information can be collected, such as the average CPU load, or the average number of active processes.

Figure 2-17. Data collection and information aggregation in Astrolabe.

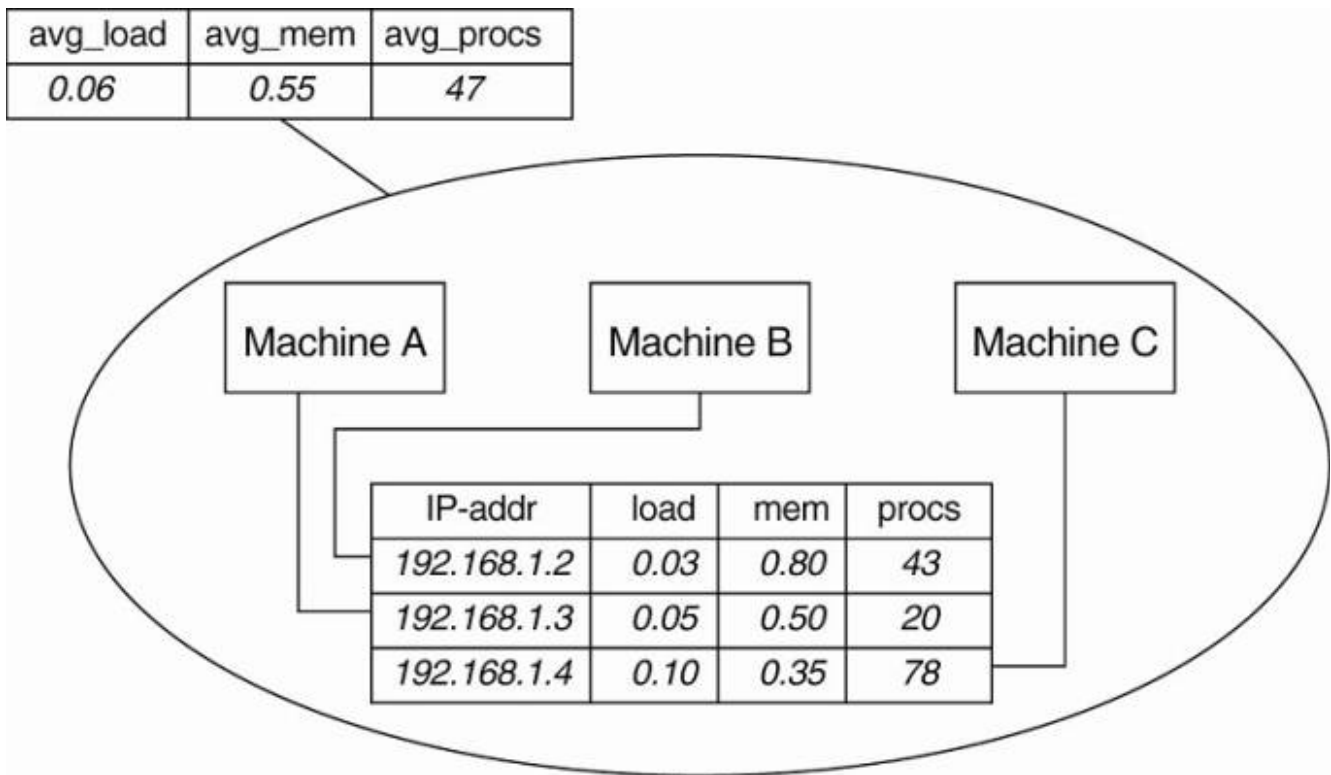


Fig. 2-17 shows how the information as gathered by each machine can be viewed as a record in a database, and that these records jointly form a relation (table). This representation is done on purpose: it is the way that Astrolabe views all the collected data. However, per zone information can only be computed from the basic records as maintained by hosts.

Aggregated information is obtained by programmable aggregation functions, which are very similar to functions available in the relational database language SQL. For example, assuming that the host information from Fig. 2-17 is maintained in a local table called `hostinfo`, we could collect the average number of processes for the zone containing machines A, B, and C, through the simple SQL query

```
SELECT AVG(procs) AS avg_procs FROM hostinfo
```

Combined with a few enhancements to SQL, it is not hard to imagine that more informative queries can be formulated.

Queries such as these are continuously evaluated by each agent running on each host. Obviously, this is possible only if zone information is propagated to all nodes that comprise Astrolabe. To this end, an agent running on a host is responsible for computing parts of the tables of its associated zones. Records for which it holds no computational responsibility are occasionally sent to it through a simple, yet effective exchange procedure known as gossiping. Gossiping protocols will be discussed in detail in Chap. 4. Likewise, an agent will pass computed results to other agents as well.

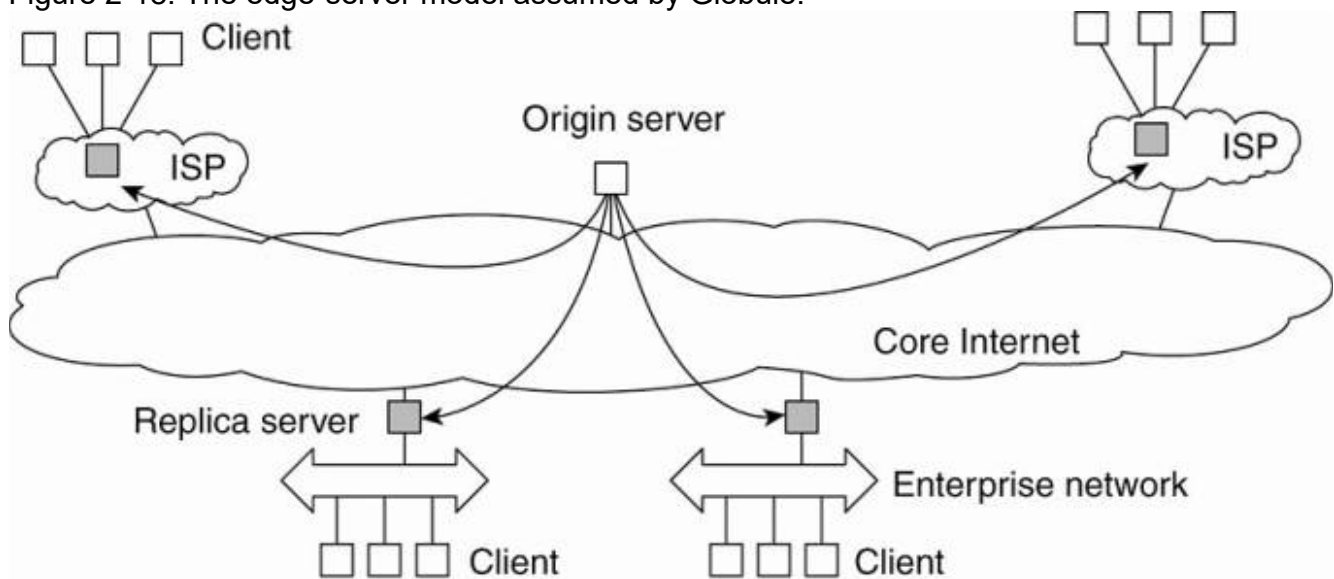
The result of this information exchange is that eventually, all agents that needed to assist in obtaining some aggregated information will see the same result (provided that no changes occur in the meantime).

2.4.3. Example: Differentiating Replication Strategies in Globule

Let us now take a look at Globule, a collaborative content distribution network (Pierre and van Steen, 2006). Globule relies on end-user servers being placed in the Internet, and that these servers collaborate to optimize performance through replication of Web pages. To this end, each origin server (i.e., the server responsible for handling updates of a specific Web site), keeps track of access patterns on a per-page basis. Access patterns are expressed as read and write operations for a page, each operation being timestamped and logged by the origin server for that page.

In its simplest form, Globule assumes that the Internet can be viewed as an edge-server system as we explained before. In particular, it assumes that requests can always be passed through an appropriate edge server, as shown in Fig. 2-18. This simple model allows an origin server to see what would have happened if it had placed a replica on a specific edge server. On the one hand, placing a replica closer to clients would improve client-perceived latency, but this will induce traffic between the origin server and that edge server in order to keep a replica consistent with the original page.

Figure 2-18. The edge-server model assumed by Globule.



When an origin server receives a request for a page, it records the IP address from where the request originated, and looks up the ISP or enterprise network associated with that request using the WHOIS Internet service (Deutsch et al., 1995). The origin server then looks for the nearest existing replica server that could act as edge server for that client, and subsequently computes the latency to that server along with the maximal bandwidth. In its simplest

configuration, Globule assumes that the latency between the replica server and the requesting user machine is negligible, and likewise that bandwidth between the two is plentiful.
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Once enough requests for a page have been collected, the origin server performs a simple "what-if analysis." Such an analysis boils down to evaluating several replication policies, where a policy describes where a specific page is replicated to, and how that page is kept consistent. Each replication policy incurs a cost that can be expressed as a simple linear function:

$$\text{cost} = (w_1 \times m_1) + (w_2 \times m_2) + \dots + (w_n \times m_n)$$

where m_k denotes a performance metric and w_k is the weight indicating how important that metric is. Typical performance metrics are the aggregated delays between a client and a replica server when returning copies of Web pages, the total consumed bandwidth between the origin server and a replica server for keeping a replica consistent, and the number of stale copies that are (allowed to be) returned to a client (Pierre et al., 2002).

For example, assume that the typical delay between the time a client C issues a request and when that page is returned from the best replica server is d_C ms. Note that what the best replica server is, is determined by a replication policy. Let m_1 denote the aggregated delay over a given time period, that is, $m_1 = \sum d_C$. If the origin server wants to optimize client-perceived latency, it will choose a relatively high value for w_1 . As a consequence, only those policies that actually minimize m_1 will show to have relatively low costs.

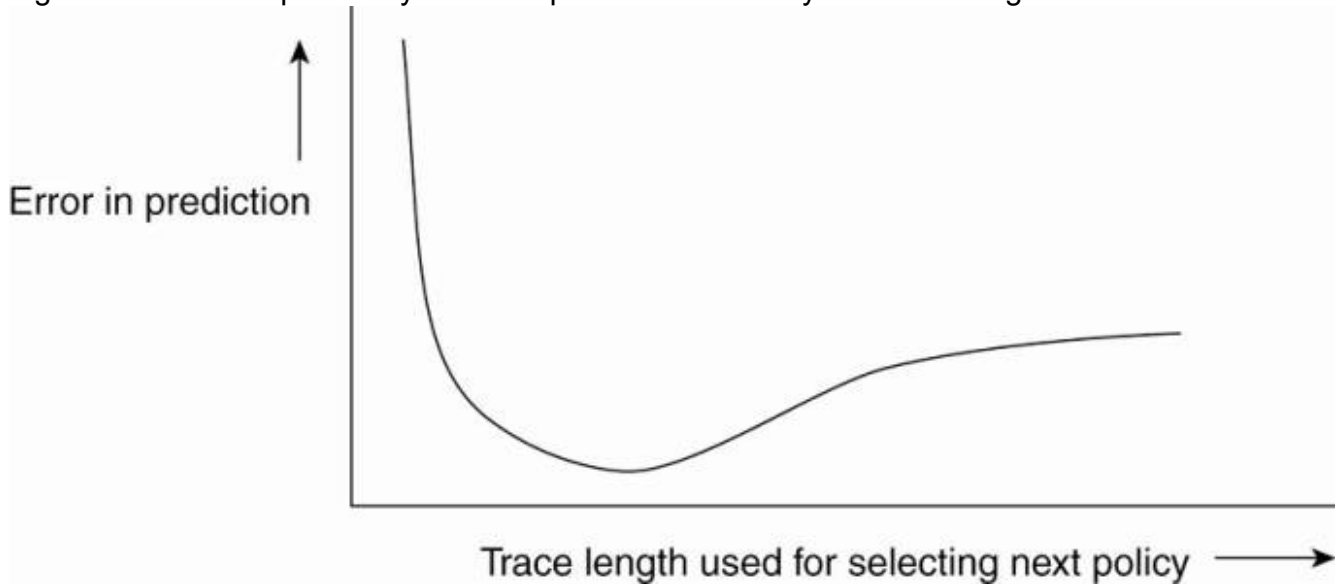
In Globule, an origin server regularly evaluates a few tens of replication policies using a trace-driven simulation, for each Web page separately. From these simulations, a best policy is selected and subsequently enforced. This may imply that new replicas are installed at different edge servers, or that a different way of keeping replicas consistent is chosen. The collecting of traces, the evaluation of replication policies, and the enforcement of a selected policy is all done automatically.

There are a number of subtle issues that need to be dealt with. For one thing, it is unclear how many requests need to be collected before an evaluation of the current policy can take place. To explain, suppose that at time T_i the origin server selects policy p for the next period until T_{i+1} . This selection takes place based on a series of past requests that were issued between T_{i-1} and T_i . Of course, in hindsight at time T_{i+1} , the server may come to the conclusion that it should have selected policy p^* given the actual requests that were issued between T_i and T_{i+1} . If p^* is different from p , then the selection of p at T_i was wrong.

As it turns out, the percentage of wrong predictions is dependent on the length of the series of requests (called the trace length) that are used to predict and select a next policy. This dependency is sketched in Fig. 2-19. What is seen is that the error in predicting the best policy goes up if the trace is not long enough. This is easily explained by the fact that we need enough requests to do a proper evaluation. However, the error also increases if we use too many requests. The reason for this is that a very long trace length captures so many changes in access patterns that predicting the best policy to follow becomes difficult, if not impossible. This phenomenon is well known and is analogous to trying to predict the weather for tomorrow by looking at what happened during the immediately preceding 100 years. A much better prediction can be made by just looking only at the recent past.

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Figure 2-19. The dependency between prediction accuracy and trace length.



Finding the optimal trace length can be done automatically as well. We leave it as an exercise to sketch a solution to this problem.

2.4.4. Example: Automatic Component Repair Management in Jade

When maintaining clusters of computers, each running sophisticated servers, it becomes important to alleviate management problems. One approach that can be applied to servers that are built using a component-based approach, is to detect component failures and have them automatically replaced. The Jade system follows this approach (Bouchenak et al., 2005). We describe it briefly in this section.

Jade is built on the Fractal component model, a Java implementation of a framework that allows components to be added and removed at runtime (Bruneton et al., 2004). A component in Fractal can have two types of interfaces. A server interface is used to call methods that are implemented by that component. A client interface is used by a component to call other components. Components are connected to each other by binding interfaces. For example, a client interface of component C1 can be bound to the server interface of component C2. A primitive binding means that a call to a client interface directly leads to calling the bounded server interface. In the case of composite binding, the call may proceed through one or more other components, for example, because the client and server interface did not match and some kind of conversion is needed. Another reason may be that the connected components lie on different machines.

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Jade uses the notion of a repair management domain. Such a domain consists of a number of nodes, where each node represents a server along with the components that are executed by that server. There is a separate node manager which is responsible for adding and removing nodes from the domain. The node manager may be replicated for assuring high availability.

Each node is equipped with failure detectors, which monitor the health of a node or one of its components and report any failures to the node manager. Typically, these detectors consider exceptional changes in the state of component, the usage of resources, and the actual failure of a component. Note that the latter may actually mean that a machine has crashed.

When a failure has been detected, a repair procedure is started. Such a procedure is driven by a repair policy, partly executed by the node manager. Policies are stated explicitly and are carried out depending on the detected failure. For example, suppose a node failure has been detected. In that case, the repair policy may prescribe that the following steps are to be carried out:

1. Terminate every binding between a component on a nonfaulty node, and a component on the node that just failed.
2. Request the node manager to start and add a new node to the domain.
3. Configure the new node with exactly the same components as those on the crashed node.
4. Re-establish all the bindings that were previously terminated.

In this example, the repair policy is simple and will only work when no crucial data has been lost (the crashed components are said to be stateless).

The approach followed by Jade is an example of self-management: upon the detection of a failure, a repair policy is automatically executed to bring the system as a whole into a state in which it was before the crash. Being a component-based system, this automatic repair requires specific support to allow components to be added and removed at runtime. In general, turning legacy applications into self-managing systems is not possible.