Chapter 4. Communication

Interprocess communication is at the heart of all distributed systems. It makes no sense to study distributed systems without carefully examining the ways that processes on different machines can exchange information. Communication in distributed systems is always based on low-level message passing as offered by the underlying network. Expressing communication through message passing is harder than using primitives based on shared memory, as available for nondistributed platforms. Modern distributed systems often consist of thousands or even millions of processes scattered across a network with unreliable communication such as the Internet. Unless the primitive communication facilities of computer networks are replaced by something else, development of large-scale distributed applications is extremely difficult.

In this chapter, we start by discussing the rules that communicating processes must adhere to, known as protocols, and concentrate on structuring those protocols in the form of layers. We then look at three widely-used models for communication: Remote Procedure Call (RPC), Message-Oriented Middleware (MOM), and data streaming. We also discuss the general problem of sending data to multiple receivers, called multicasting.

Our first model for communication in distributed systems is the remote procedure call (RPC). An RPC aims at hiding most of the intricacies of message passing, and is ideal for client-server applications.

In many distributed applications, communication does not follow the rather strict pattern of client-server interaction. In those cases, it turns out that thinking in terms of messages is more appropriate. However, the low-level communication facilities of computer networks are in many ways not suitable due to their lack of distribution transparency. An alternative is to use a high-level message-queuing model, in which communication proceeds much the same as in electronic mail systems. Message-oriented middleware (MOM) is a subject important enough to warrant a section of its own.
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With the advent of multimedia distributed systems, it became apparent that many systems were lacking support for communication of continuous media, such as audio and video. What is needed is the notion of a stream that can support the continuous flow of messages, subject to various timing constraints. Streams are discussed in a separate section.

Finally, since our understanding of setting up multicast facilities has improved, novel and elegant solutions for data dissemination have emerged. We pay separate attention to this subject in the last section of this chapter.

4.1. Fundamentals

Before we start our discussion on communication in distributed systems, we first recapitulate some of the fundamental issues related to communication. In the next section we briefly discuss network communication protocols, as these form the basis for any distributed system. After that, we take a different approach by classifying the different types of communication that occurs in distributed systems.

4.1.1. Layered Protocols

Due to the absence of shared memory, all communication in distributed systems is based on sending and receiving (low level) messages. When process A wants to communicate with process B, it first builds a message in its own address space. Then it executes a system call that causes the operating system to send the message over the network to B. Although this basic idea sounds simple enough, in order to prevent chaos, A and B have to agree on the meaning of the bits being sent. If A sends a brilliant new novel written in French and encoded in IBM's EBCDIC character code, and B expects the inventory of a supermarket written in English and encoded in ASCII, communication will be less than optimal.

Many different agreements are needed. How many volts should be used to signal a 0-bit, and how many volts for a 1-bit? How does the receiver know which is the last bit of the message? How can it detect if a message has been damaged or lost, and what should it do if it finds out? How long are numbers, strings, and other data items, and how are they represented? In short, agreements are needed at a variety of levels, varying from the low-level details of bit transmission to the high-level details of how information is to be expressed.
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To make it easier to deal with the numerous levels and issues involved in communication, the International Standards Organization (ISO) developed a reference model that clearly identifies the various levels involved, gives them standard names, and points out which level should do which job. This model is called the Open Systems Interconnection Reference Model (Day and Zimmerman, 1983), usually abbreviated as ISO OSI or sometimes just the OSI model. It should be emphasized that the protocols that were developed as part of the OSI model were never widely used and are essentially dead now. However, the underlying model itself has proved to be quite useful for understanding computer networks. Although we do not intend to give a full description of this model and all of its implications here, a short introduction will be helpful. For more details, see Tanen-baum (2003).

The OSI model is designed to allow open systems to communicate. An open system is one that is prepared to communicate with any other open system by using standard rules that govern the format, contents, and meaning of the messages sent and received. These rules are formalized in what are called protocols. To allow a group of computers to communicate over a network, they must all agree on the protocols to be used. A distinction is made between two general types of protocols. With connection oriented protocols, before exchanging data the sender and

receiver first explicitly establish a connection, and possibly negotiate the protocol they will use. When they are done, they must release (terminate) the connection. The telephone is a connection-oriented communication system. With connectionless protocols, no setup in advance is needed. The sender just transmits the first message when it is ready. Dropping a letter in a mailbox is an example of connectionless communication. With computers, both connection-oriented and connectionless communication are common.

In the OSI model, communication is divided up into seven levels or layers, as shown in Fig. 4-1. Each layer deals with one specific aspect of the communication. In this way, the problem can be divided up into manageable pieces, each of which can be solved independent of the others. Each layer provides an interface to the one above it. The interface consists of a set of operations that together define the service the layer is prepared to offer its users.

Figure 4-1. Layers, interfaces, and protocols in the OSI model.
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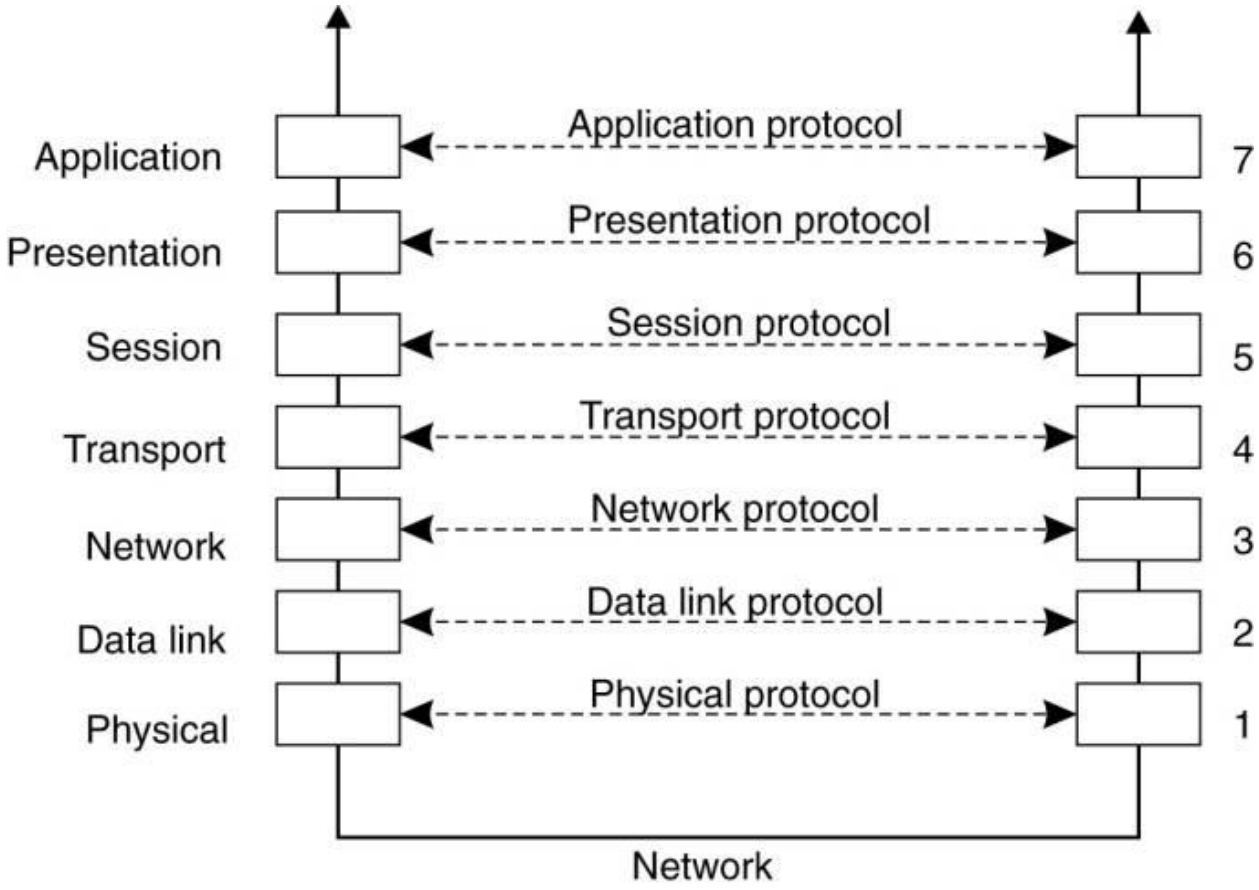

When process A on machine 1 wants to communicate with process B on machine 2, it builds a message and passes the message to the application layer on its machine. This layer might be a

library procedure, for example, but it could also be implemented in some other way (e.g., inside the operating system, on an external network processor, etc.). The application layer software then adds a header to the front of the message and passes the resulting message across the layer 6/7 interface to the presentation layer. The presentation layer in turn adds its own header and passes the result down to the session layer, and so on. Some layers add not only a header to the front, but also a trailer to the end. When it hits the bottom, the physical layer actually transmits the message (which by now might look as shown in Fig. 4-2) by putting it onto the physical transmission medium.
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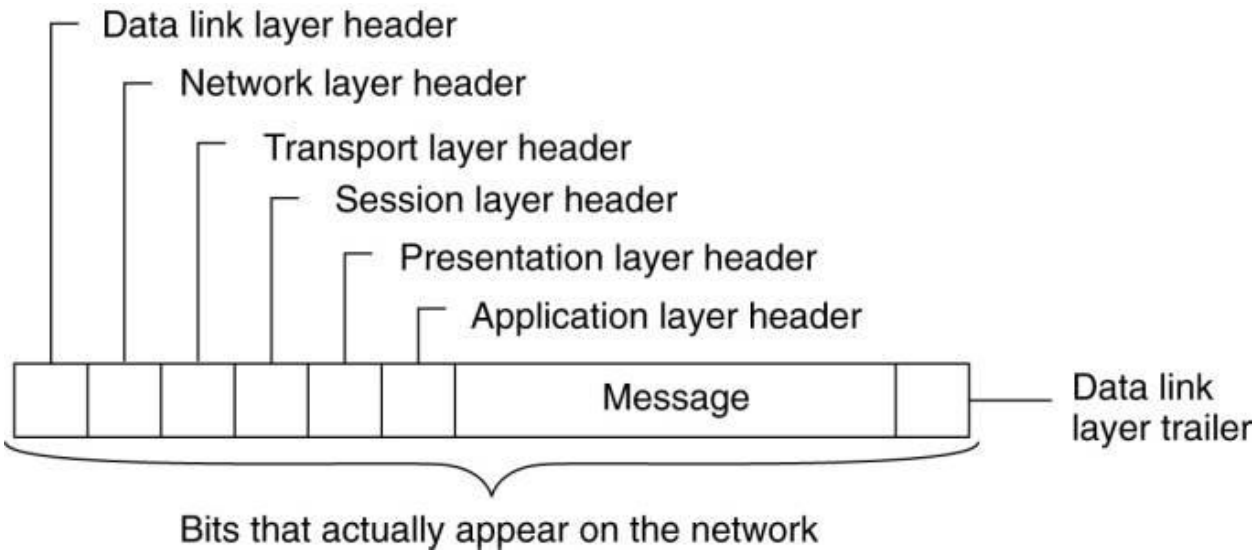
Figure 4-2. A typical message as it appears on the network.



When the message arrives at machine 2, it is passed upward, with each layer stripping off and examining its own header. Finally, the message arrives at the receiver, process B, which may reply to it using the reverse path. The information in the layer n header is used for the layer n protocol.

As an example of why layered protocols are important, consider communication between two companies, Zippy Airlines and its caterer, Mushy Meals, Inc. Every month, the head of passenger service at Zippy asks her secretary to contact the sales manager's secretary at Mushy to order 100,000 boxes of rubber chicken. Traditionally, the orders went via the post office. However, as the postal service deteriorated, at some point the two secretaries decided to abandon it and communicate by e-mail. They could do this without bothering their bosses, since their protocol deals with the physical transmission of the orders, not their contents.
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Similarly, the head of passenger service can decide to drop the rubber chicken and go for Mushy's new special, prime rib of goat, without that decision affecting the secretaries. The thing to notice is that we have two layers here, the bosses and the secretaries. Each layer has its own protocol (subjects of discussion and technology) that can be changed independently of the other one. It is precisely this independence that makes layered protocols attractive. Each one can be changed as technology improves, without the other ones being affected.

In the OSI model, there are not two layers, but seven, as we saw in Fig. 4-1. The collection of protocols used in a particular system is called a protocol suite or protocol stack. It is important to distinguish a reference model from its actual protocols. As we mentioned, the OSI protocols were never popular. In contrast, protocols developed for the Internet, such as TCP and IP, are mostly used. In the following sections, we will briefly examine each of the OSI layers in turn, starting at the bottom. However, instead of giving examples of OSI protocols, where appropriate, we will point out some of the Internet protocols used in each layer.

Lower-Level Protocols

We start with discussing the three lowest layers of the OSI protocol suite. Together, these layers implement the basic functions that encompass a computer network.

The physical layer is concerned with transmitting the 0s and 1s. How many volts to use for 0 and 1, how many bits per second can be sent, and whether transmission can take place in both directions simultaneously are key issues in the physical layer. In addition, the size and shape of the network connector (plug), as well as the number of pins and meaning of each are of concern here.

The physical layer protocol deals with standardizing the electrical, mechanical, and signaling interfaces so that when one machine sends a 0 bit it is actually received as a 0 bit and not a 1 bit. Many physical layer standards have been developed (for different media), for example, the RS-232-C standard for serial communication lines.

The physical layer just sends bits. As long as no errors occur, all is well. However, real communication networks are subject to errors, so some mechanism is needed to detect and correct them. This mechanism is the main task of the data link layer. What it does is to group the bits into units, sometimes called frames, and see that each frame is correctly received.

The data link layer does its work by putting a special bit pattern on the start and end of each frame to mark them, as well as computing a checksum by adding up all the bytes in the frame in a certain way. The data link layer appends the checksum to the frame. When the frame arrives, the receiver recomputes the checksum from the data and compares the result to the checksum following the frame. If the two agree, the frame is considered correct and is accepted. It they

disagree, the receiver asks the sender to retransmit it. Frames are assigned sequence numbers (in the header), so everyone can tell which is which.
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On a LAN, there is usually no need for the sender to locate the receiver. It just puts the message out on the network and the receiver takes it off. A wide-area network, however, consists of a large number of machines, each with some number of lines to other machines, rather like a large-scale map showing major cities and roads connecting them. For a message to get from the sender to the receiver it may have to make a number of hops, at each one choosing an outgoing line to use. The question of how to choose the best path is called routing, and is essentially the primary task of the network layer.

The problem is complicated by the fact that the shortest route is not always the best route. What really matters is the amount of delay on a given route, which, in turn, is related to the amount of traffic and the number of messages queued up for transmission over the various lines. The delay can thus change over the course of time. Some routing algorithms try to adapt to changing loads, whereas others are content to make decisions based on long-term averages.

At present, the most widely used network protocol is the connectionless IP (Internet Protocol), which is part of the Internet protocol suite. An IP packet (the technical term for a message in the network layer) can be sent without any setup. Each IP packet is routed to its destination independent of all others. No internal path is selected and remembered.

Transport Protocols

The transport layer forms the last part of what could be called a basic network protocol stack, in the sense that it implements all those services that are not provided at the interface of the network layer, but which are reasonably needed to build network applications. In other words, the transport layer turns the underlying network into something that an application developer can use.

Packets can be lost on the way from the sender to the receiver. Although some applications can handle their own error recovery, others prefer a reliable connection. The job of the transport layer is to provide this service. The idea is that the application layer should be able to deliver a message to the transport layer with the expectation that it will be delivered without loss.

Upon receiving a message from the application layer, the transport layer breaks it into pieces small enough for transmission, assigns each one a sequence number, and then sends them all. The discussion in the transport layer header concerns which packets have been sent, which have been received, how many more the receiver has room to accept, which should be retransmitted, and similar topics.

Reliable transport connections (which by definition are connection oriented) can be built on top of connection-oriented or connectionless network services. In the former case all the packets will arrive in the correct sequence (if they arrive at all), but in the latter case it is possible for one packet to take a different route and arrive earlier than the packet sent before it. It is up to the transport layer software to put everything back in order to maintain the illusion that a transport connection is like a big tube—you put messages into it and they come out undamaged and in the same order in which they went in. Providing this end-to-end communication behavior is an important aspect of the transport layer.
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The Internet transport protocol is called TCP (Transmission Control Protocol) and is described in detail in Comer (2006). The combination TCP/IP is now used as a de facto standard for network communication. The Internet protocol suite also supports a connectionless transport protocol called UDP (Universal Datagram Protocol), which is essentially just IP with some minor additions. User programs that do not need a connection-oriented protocol normally use UDP.

Additional transport protocols are regularly proposed. For example, to support real-time data transfer, the Real-time Transport Protocol (RTP) has been defined. RTP is a framework protocol in the sense that it specifies packet formats for real-time data without providing the actual mechanisms for guaranteeing data delivery. In addition, it specifies a protocol for monitoring and controlling data transfer of RTP packets (Schulzrinne et al., 2003).

Higher-Level Protocols

Above the transport layer, OSI distinguished three additional layers. In practice, only the application layer is ever used. In fact, in the Internet protocol suite, everything above the transport layer is grouped together. In the face of middle-ware systems, we shall see in this section that neither the OSI nor the Internet approach is really appropriate.

The session layer is essentially an enhanced version of the transport layer. It provides dialog control, to keep track of which party is currently talking, and it provides synchronization facilities. The latter are useful to allow users to insert checkpoints into long transfers, so that in the event of a crash, it is necessary to go back only to the last checkpoint, rather than all the way back to the beginning. In practice, few applications are interested in the session layer and it is rarely supported. It is not even present in the Internet protocol suite. However, in the context of developing middleware solutions, the concept of a session and its related protocols has turned out to be quite relevant, notably when defining higher-level communication protocols.

Unlike the lower layers, which are concerned with getting the bits from the sender to the receiver reliably and efficiently, the presentation layer is concerned with the meaning of the bits. Most messages do not consist of random bit strings, but more structured information such as people's names, addresses, amounts of money, and so on. In the presentation layer it is possible to define records containing fields like these and then have the sender notify the

receiver that a message contains a particular record in a certain format. This makes it easier for machines with different internal representations to communicate with each other.
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The OSI application layer was originally intended to contain a collection of standard network applications such as those for electronic mail, file transfer, and terminal emulation. By now, it has become the container for all applications and protocols that in one way or the other do not fit into one of the underlying layers. From the perspective of the OSI reference model, virtually all distributed systems are just applications.

What is missing in this model is a clear distinction between applications, application-specific protocols, and general-purpose protocols. For example, the Internet File Transfer Protocol (FTP) (Postel and Reynolds, 1985; and Horowitz and Lunt, 1997) defines a protocol for transferring files between a client and server machine. The protocol should not be confused with the ftp program, which is an end-user application for transferring files and which also (not entirely by coincidence) happens to implement the Internet FTP.

Another example of a typical application-specific protocol is the HyperText Transfer Protocol (HTTP) (Fielding et al., 1999), which is designed to remotely manage and handle the transfer of Web pages. The protocol is implemented by applications such as Web browsers and Web servers. However, HTTP is now also used by systems that are not intrinsically tied to the Web. For example, Java's object-invocation mechanism uses HTTP to request the invocation of remote objects that are protected by a firewall (Sun Microsystems, 2004b).

There are also many general-purpose protocols that are useful to many applications, but which cannot be qualified as transport protocols. In many cases, such protocols fall into the category of middleware protocols, which we discuss next.

Middleware Protocols

Middleware is an application that logically lives (mostly) in the application layer, but which contains many general-purpose protocols that warrant their own layers, independent of other, more specific applications. A distinction can be made between high-level communication protocols and protocols for establishing various middleware services.

There are numerous protocols to support a variety of middleware services. For example, as we discuss in Chap. 9, there are various ways to establish authentication, that is, provide proof of a claimed identity. Authentication protocols are not closely tied to any specific application, but instead, can be integrated into a middleware system as a general service. Likewise, authorization protocols by which authenticated users and processes are granted access only to those resources for which they have authorization, tend to have a general, application-independent nature.

As another example, we shall consider a number of distributed commit protocols in Chap. 8. Commit protocols establish that in a group of processes either all processes carry out a particular operation, or that the operation is not carried out at all. This phenomenon is also referred to as atomicity and is widely applied in transactions. As we shall see, besides transactions, other applications, like fault-tolerant ones, can also take advantage of distributed commit protocols.
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As a last example, consider a distributed locking protocol by which a resource can be protected against simultaneous access by a collection of processes that are distributed across multiple machines. We shall come across a number of such protocols in Chap. 6. Again, this is an example of a protocol that can be used to implement a general middleware service, but which, at the same time, is highly independent of any specific application.

Middleware communication protocols support high-level communication services. For example, in the next two sections we shall discuss protocols that allow a process to call a procedure or invoke an object on a remote machine in a highly transparent way. Likewise, there are high-level communication services for setting and synchronizing streams for transferring real-time data, such as needed for multimedia applications. As a last example, some middleware systems offer reliable multicast services that scale to thousands of receivers spread across a wide-area network.

Some of the middleware communication protocols could equally well belong in the transport layer, but there may be specific reasons to keep them at a higher level. For example, reliable multicasting services that guarantee scalability can be implemented only if application requirements are taken into account. Consequently, a middleware system may offer different (tunable) protocols, each in turn implemented using different transport protocols, but offering a single interface.

Taking this approach to layering leads to a slightly adapted reference model for communication, as shown in Fig. 4-3. Compared to the OSI model, the session and presentation layer have been replaced by a single middleware layer that contains application-independent protocols. These protocols do not belong in the lower layers we just discussed. The original transport services may also be offered as a middleware service, without being modified. This approach is somewhat analogous to offering UDP at the transport level. Likewise, middleware communication services may include message-passing services comparable to those offered by the transport layer.
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Figure 4-3. An adapted reference model for networked communication.
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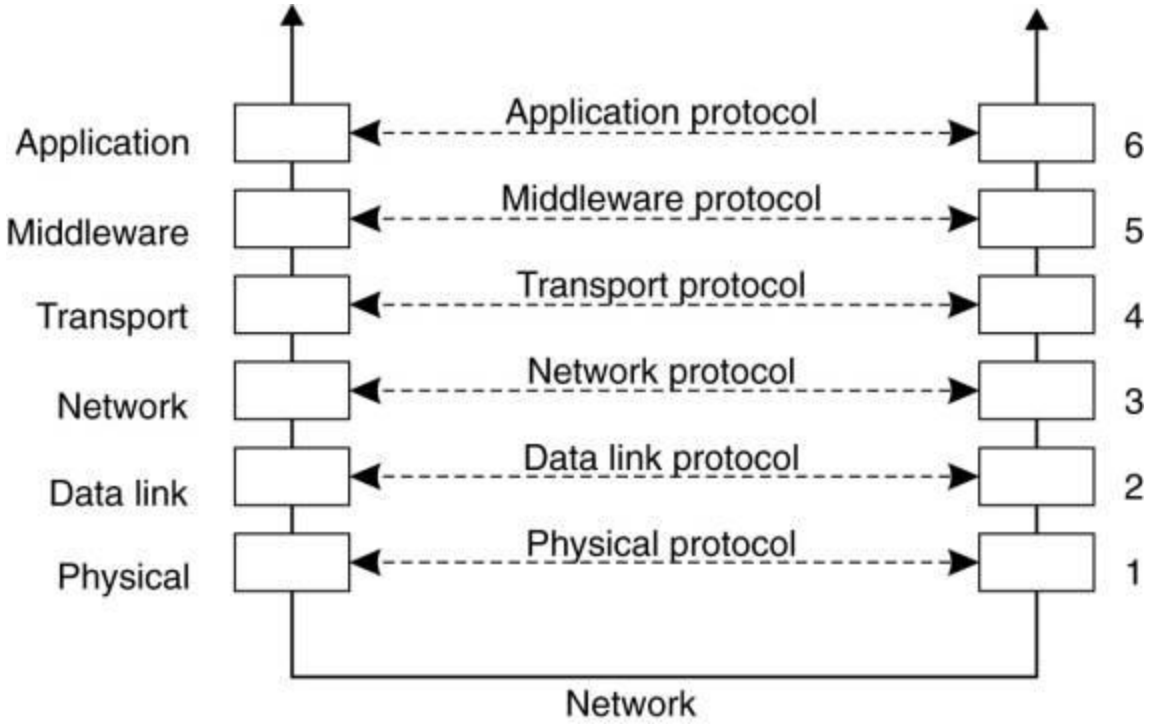
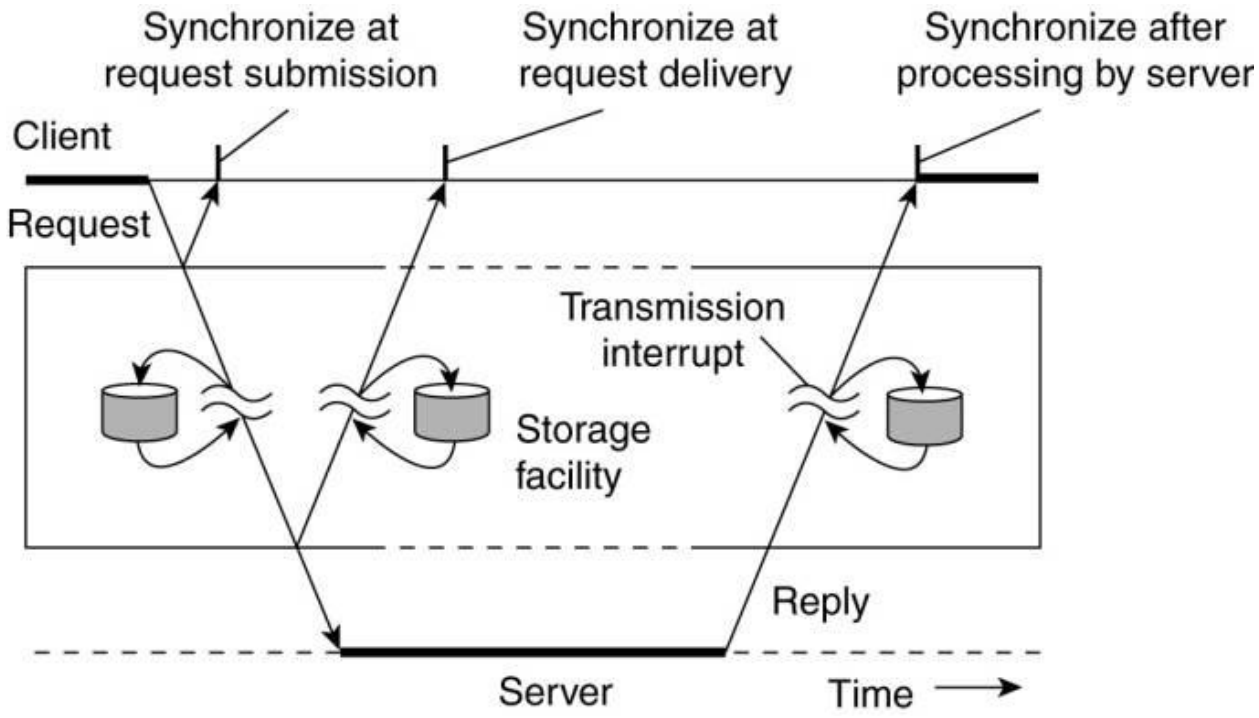


In the remainder of this chapter, we concentrate on four high-level middle-ware communication services: remote procedure calls, message queuing services, support for communication of continuous media through streams, and multicasting. Before doing so, there are other general criteria for distinguishing (middleware) communication which we discuss next.

4.1.2. Types of Communication

To understand the various alternatives in communication that middleware can offer to applications, we view the middleware as an additional service in client-server computing, as shown in Fig. 4-4. Consider, for example an electronic mail system. In principle, the core of the mail delivery system can be seen as a middleware communication service. Each host runs a user agent allowing users to compose, send, and receive e-mail. A sending user agent passes such mail to the mail delivery system, expecting it, in turn, to eventually deliver the mail to the intended recipient. Likewise, the user agent at the receiver's side connects to the mail delivery system to see whether any mail has come in. If so, the messages are transferred to the user agent so that they can be displayed and read by the user.

Figure 4-4. Viewing middleware as an intermediate (distributed) service in application-level communication.



An electronic mail system is a typical example in which communication is persistent. With persistent communication, a message that has been submitted for transmission is stored by the communication middleware as long as it takes to deliver it to the receiver. In this case, the middleware will store the message at one or several of the storage facilities shown in Fig. 4-4. As a consequence, it is not necessary for the sending application to continue execution after submitting the message. Likewise, the receiving application need not be executing when the message is submitted.
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In contrast, with transient communication, a message is stored by the communication system only as long as the sending and receiving application are executing. More precisely, in terms of Fig. 4-4, the middleware cannot deliver a message due to a transmission interrupt, or because the recipient is currently not active, it will simply be discarded. Typically, all transport-level communication services offer only transient communication. In this case, the communication system consists traditional store-and-forward routers. If a router cannot deliver a message to the next one or the destination host, it will simply drop the message.

Besides being persistent or transient, communication can also be asynchronous or synchronous. The characteristic feature of asynchronous communication is that a sender continues immediately after it has submitted its message for transmission. This means that the message is (temporarily) stored immediately by the middleware upon submission. With

synchronous communication, the sender is blocked until its request is known to be accepted. There are essentially three points where synchronization can take place. First, the sender may be blocked until the middleware notifies that it will take over transmission of the request. Second, the sender may synchronize until its request has been delivered to the intended recipient. Third, synchronization may take place by letting the sender wait until its request has been fully processed, that is, up the time that the recipient returns a response.

Various combinations of persistence and synchronization occur in practice. Popular ones are persistence in combination with synchronization at request submission, which is a common scheme for many message-queuing systems, which we discuss later in this chapter. Likewise, transient communication with synchronization after the request has been fully processed is also widely used. This scheme corresponds with remote procedure calls, which we also discuss below.

Besides persistence and synchronization, we should also make a distinction between discrete and streaming communication. The examples so far all fall in the category of discrete communication: the parties communicate by messages, each message forming a complete unit of information. In contrast, streaming involves sending multiple messages, one after the other, where the messages are related to each other by the order they are sent, or because there is a temporal relationship. We return to streaming communication extensively below.

4.2. Remote Procedure Call

Many distributed systems have been based on explicit message exchange between processes. However, the procedures send and receive do not conceal communication at all, which is important to achieve access transparency in distributed systems. This problem has long been known, but little was done about it until a paper by Birrell and Nelson (1984) introduced a completely different way of handling communication. Although the idea is refreshingly simple (once someone has thought of it), the implications are often subtle. In this section we will examine the concept, its implementation, its strengths, and its weaknesses.
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In a nutshell, what Birrell and Nelson suggested was allowing programs to call procedures located on other machines. When a process on machine A calls a procedure on machine B, the calling process on A is suspended, and execution of the called procedure takes place on B. Information can be transported from the caller to the callee in the parameters and can come back in the procedure result. No message passing at all is visible to the programmer. This method is known as Remote Procedure Call, or often just RPC.

While the basic idea sounds simple and elegant, subtle problems exist. To start with, because the calling and called procedures run on different machines, they execute in different address spaces, which causes complications. Parameters and results also have to be passed, which can be complicated, especially if the machines are not identical. Finally, either or both machines can

crash and each of the possible failures causes different problems. Still, most of these can be dealt with, and RPC is a widely-used technique that underlies many distributed systems.

4.2.1. Basic RPC Operation

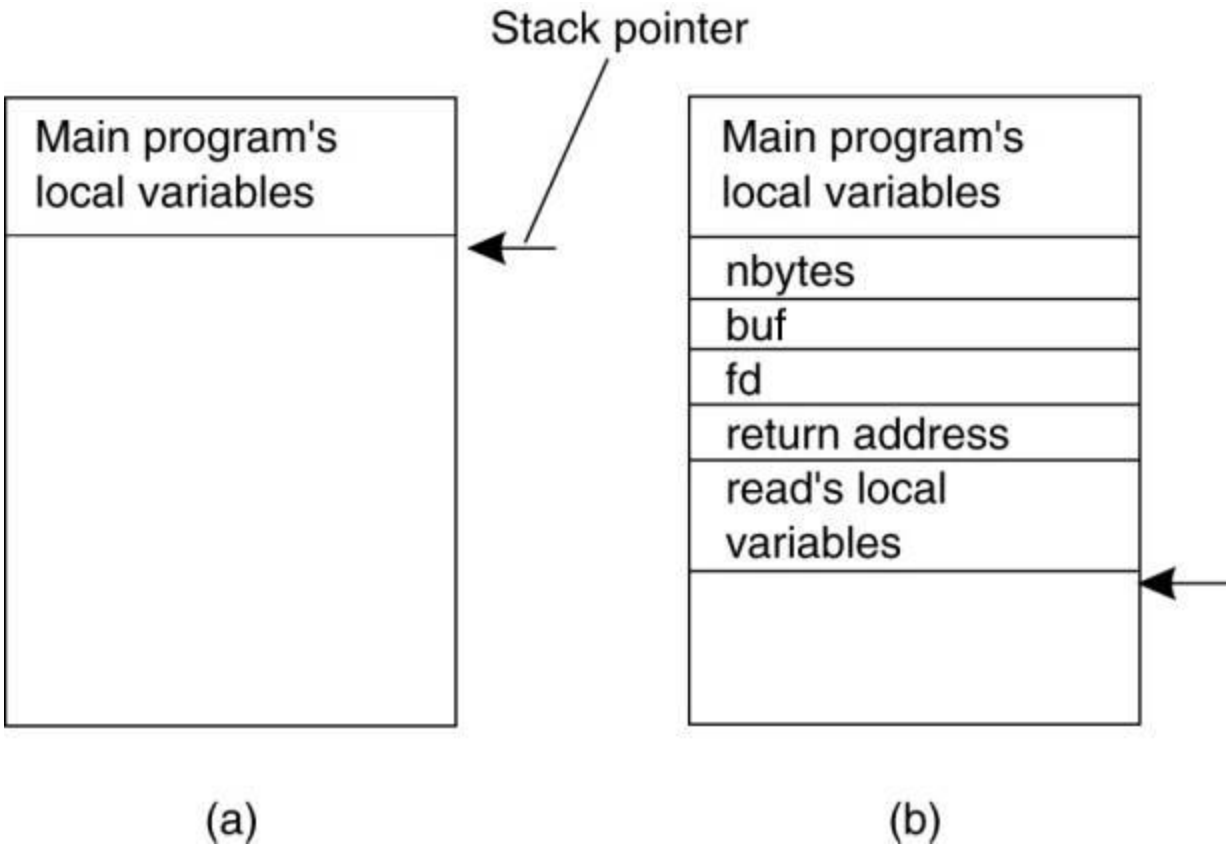
We first start with discussing conventional procedure calls, and then explain how the call itself can be split into a client and server part that are each executed on different machines.
Conventional Procedure Call

To understand how RPC works, it is important first to fully understand how a conventional (i.e., single machine) procedure call works. Consider a call in C like

```
count = read(fd, buf, nbytes);
```

where fd is an integer indicating a file, buf is an array of characters into which data are read, and nbytes is another integer telling how many bytes to read. If the call is made from the main program, the stack will be as shown in Fig. 4-5(a) before the call. To make the call, the caller pushes the parameters onto the stack in order, last one first, as shown in Fig. 4-5(b). (The reason that C compilers push the parameters in reverse order has to do with printf—by doing so, printf can always locate its first parameter, the format string.) After the read procedure has finished running, it puts the return value in a register, removes the return address, and transfers control back to the caller. The caller then removes the parameters from the stack, returning the stack to the original state it had before the call.
[Page 127]

Figure 4-5. (a) Parameter passing in a local procedure call: the stack before the call to read. (b) The stack while the called procedure is active.



Several things are worth noting. For one, in C, parameters can be call-by-value or call-by-reference. A value parameter, such as fd or nbytes, is simply copied to the stack as shown in Fig. 4-5(b). To the called procedure, a value parameter is just an initialized local variable. The called procedure may modify it, but such changes do not affect the original value at the calling side.

A reference parameter in C is a pointer to a variable (i.e., the address of the variable), rather than the value of the variable. In the call to read, the second parameter is a reference parameter because arrays are always passed by reference in C. What is actually pushed onto the stack is the address of the character array. If the called procedure uses this parameter to store something into the character array, it does modify the array in the calling procedure. The difference between call-by-value and call-by-reference is quite important for RPC, as we shall see.

One other parameter passing mechanism also exists, although it is not used in C. It is called call-by-copy/restore. It consists of having the variable copied to the stack by the caller, as in call-by-value, and then copied back after the call, overwriting the caller's original value. Under most conditions, this achieves exactly the same effect as call-by-reference, but in some

situations, such as the same parameter being present multiple times in the parameter list, the semantics are different. The call-by-copy/restore mechanism is not used in many languages.

The decision of which parameter passing mechanism to use is normally made by the language designers and is a fixed property of the language. Sometimes it depends on the data type being passed. In C, for example, integers and other scalar types are always passed by value, whereas arrays are always passed by reference, as we have seen. Some Ada compilers use copy/restore for in out parameters, but others use call-by-reference. The language definition permits either choice, which makes the semantics a bit fuzzy.
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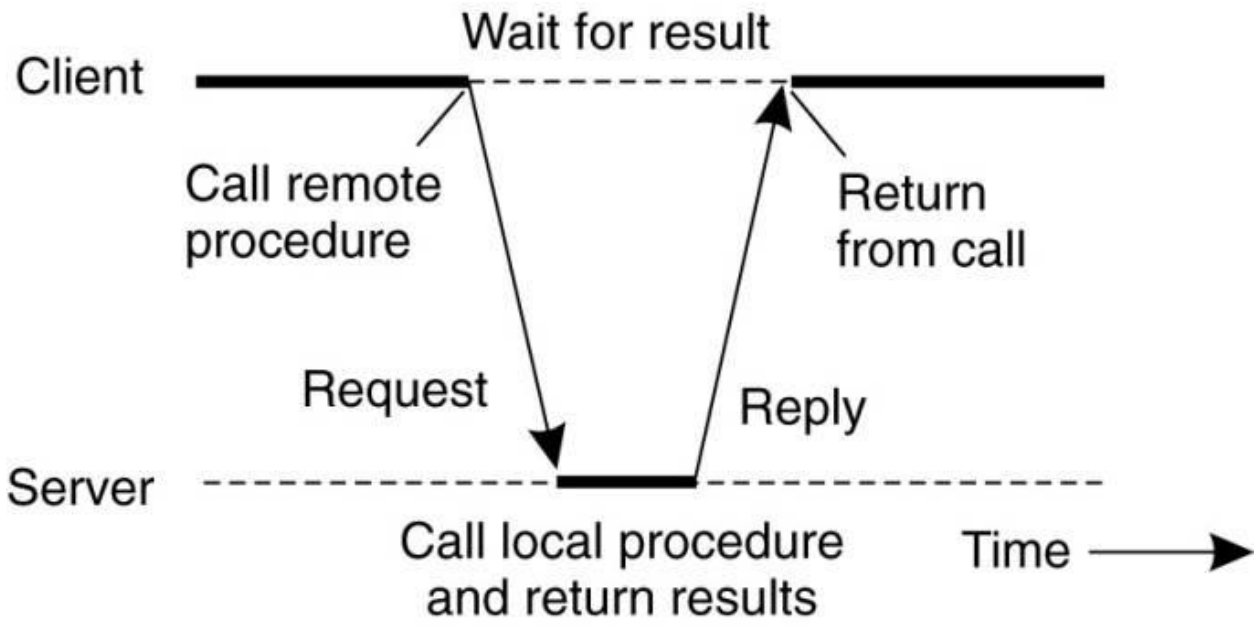
Client and Server Stubs

The idea behind RPC is to make a remote procedure call look as much as possible like a local one. In other words, we want RPC to be transparent—the calling procedure should not be aware that the called procedure is executing on a different machine or vice versa. Suppose that a program needs to read some data from a file. The programmer puts a call to read in the code to get the data. In a traditional (single-processor) system, the read routine is extracted from the library by the linker and inserted into the object program. It is a short procedure, which is generally implemented by calling an equivalent read system call. In other words, the read procedure is a kind of interface between the user code and the local operating system.

Even though read does a system call, it is called in the usual way, by pushing the parameters onto the stack, as shown in Fig. 4-5(b). Thus the programmer does not know that read is actually doing something fishy.

RPC achieves its transparency in an analogous way. When read is actually a remote procedure (e.g., one that will run on the file server's machine), a different version of read, called a client stub, is put into the library. Like the original one, it, too, is called using the calling sequence of Fig. 4-5(b). Also like the original one, it too, does a call to the local operating system. Only unlike the original one, it does not ask the operating system to give it data. Instead, it packs the parameters into a message and requests that message to be sent to the server as illustrated in Fig. 4-6. Following the call to send, the client stub calls receive, blocking itself until the reply comes back.

Figure 4-6. Principle of RPC between a client and server program.



When the message arrives at the server, the server's operating system passes it up to a server stub. A server stub is the server-side equivalent of a client stub: it is a piece of code that transforms requests coming in over the network into local procedure calls. Typically the server stub will have called receive and be blocked waiting for incoming messages. The server stub unpacks the parameters from the message and then calls the server procedure in the usual way (i.e., as in Fig. 4-5). From the server's point of view, it is as though it is being called directly by the client—the parameters and return address are all on the stack where they belong and nothing seems unusual. The server performs its work and then returns the result to the caller in the usual way. For example, in the case of read, the server will fill the buffer, pointed to by the second parameter, with the data. This buffer will be internal to the server stub.
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When the server stub gets control back after the call has completed, it packs the result (the buffer) in a message and calls send to return it to the client. After that, the server stub usually does a call to receive again, to wait for the next incoming request.

When the message gets back to the client machine, the client's operating system sees that it is addressed to the client process (or actually the client stub, but the operating system cannot see the difference). The message is copied to the waiting buffer and the client process unblocked. The client stub inspects the message, unpacks the result, copies it to its caller, and returns in the usual way. When the caller gets control following the call to read, all it knows is that its data are available. It has no idea that the work was done remotely instead of by the local operating system.

This blissful ignorance on the part of the client is the beauty of the whole scheme. As far as it is concerned, remote services are accessed by making ordinary (i.e., local) procedure calls, not by calling send and receive. All the details of the message passing are hidden away in the two library procedures, just as the details of actually making system calls are hidden away in traditional libraries.

To summarize, a remote procedure call occurs in the following steps:

1. The client procedure calls the client stub in the normal way.
2. The client stub builds a message and calls the local operating system.
3. The client's OS sends the message to the remote OS.
4. The remote OS gives the message to the server stub.
5. The server stub unpacks the parameters and calls the server.
6. The server does the work and returns the result to the stub.
7. The server stub packs it in a message and calls its local OS.
8. The server's OS sends the message to the client's OS.
9. The client's OS gives the message to the client stub.
10. The stub unpacks the result and returns to the client.

The net effect of all these steps is to convert the local call by the client procedure to the client stub, to a local call to the server procedure without either client or server being aware of the intermediate steps or the existence of the network.
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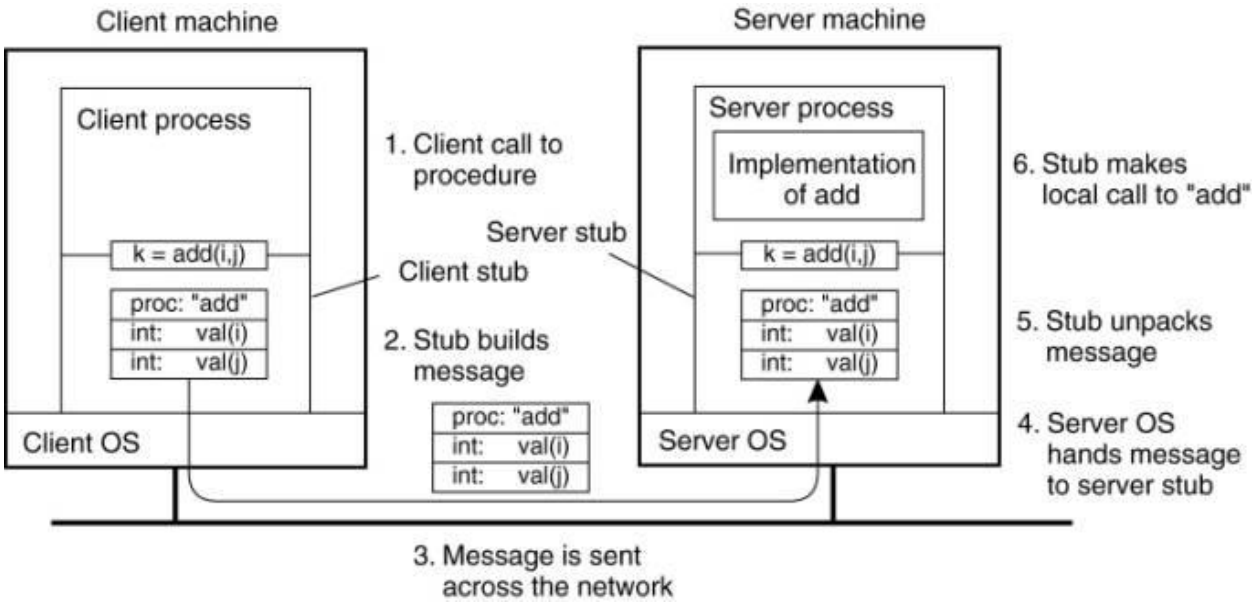
4.2.2. Parameter Passing

The function of the client stub is to take its parameters, pack them into a message, and send them to the server stub. While this sounds straightforward, it is not quite as simple as it at first appears. In this section we will look at some of the issues concerned with parameter passing in RPC systems.
Passing Value Parameters

Packing parameters into a message is called parameter marshaling. As a very simple example, consider a remote procedure, add(i, j), that takes two integer parameters i and j and returns

their arithmetic sum as a result. (As a practical matter, one would not normally make such a simple procedure remote due to the overhead, but as an example it will do.) The call to add, is shown in the left-hand portion (in the client process) in Fig. 4-7. The client stub takes its two parameters and puts them in a message as indicated. It also puts the name or number of the procedure to be called in the message because the server might support several different calls, and it has to be told which one is required.

Figure 4-7. The steps involved in a doing a remote computation through RPC.



When the message arrives at the server, the stub examines the message to see which procedure is needed and then makes the appropriate call. If the server also supports other remote procedures, the server stub might have a switch statement in it to select the procedure to be called, depending on the first field of the message. The actual call from the stub to the server looks like the original client call, except that the parameters are variables initialized from the incoming message.

When the server has finished, the server stub gains control again. It takes the result sent back by the server and packs it into a message. This message is sent back back to the client stub, which unpacks it to extract the result and returns the value to the waiting client procedure.
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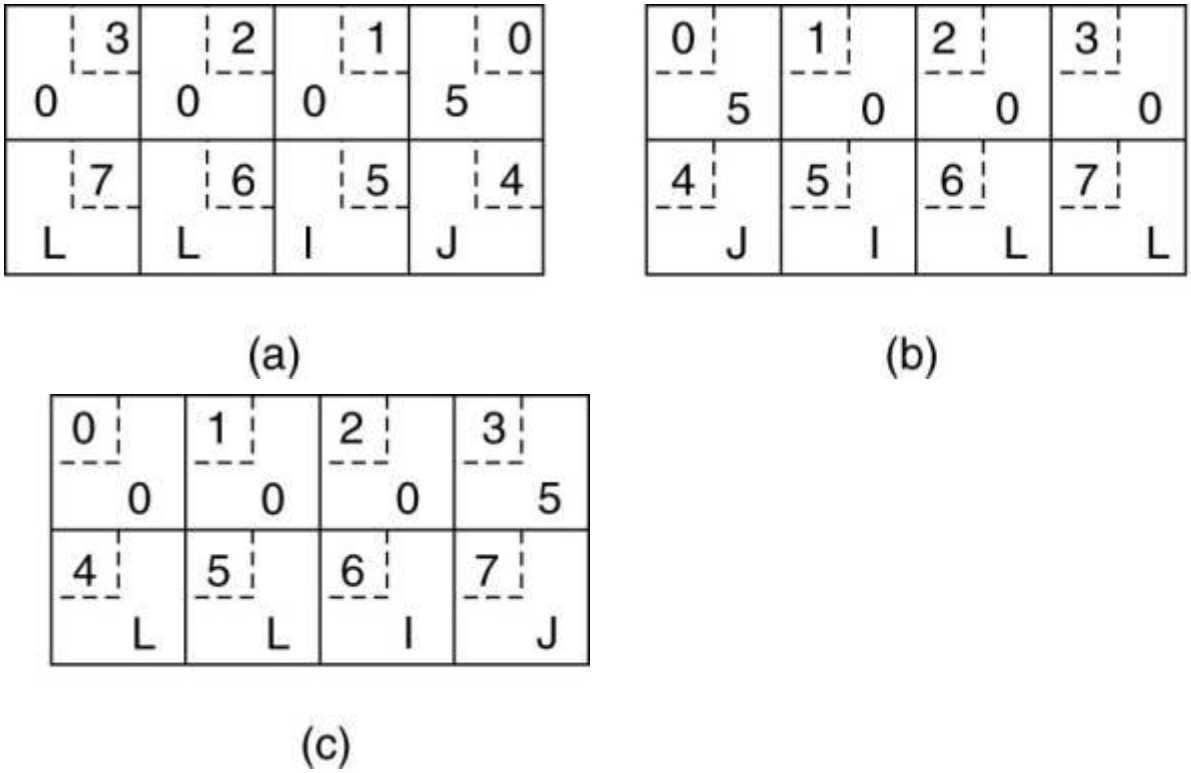
As long as the client and server machines are identical and all the parameters and results are scalar types, such as integers, characters, and Booleans, this model works fine. However, in a large distributed system, it is common that multiple machine types are present. Each machine

often has its own representation for numbers, characters, and other data items. For example, IBM mainframes use the EBCDIC character code, whereas IBM personal computers use ASCII. As a consequence, it is not possible to pass a character parameter from an IBM PC client to an IBM mainframe server using the simple scheme of Fig. 4-7: the server will interpret the character incorrectly.

Similar problems can occur with the representation of integers (one's complement versus two's complement) and floating-point numbers. In addition, an even more annoying problem exists because some machines, such as the Intel Pentium, number their bytes from right to left, whereas others, such as the Sun SPARC, number them the other way. The Intel format is called little endian and the SPARC format is called big endian, after the politicians in Gulliver's Travels who went to war over which end of an egg to break (Cohen, 1981). As an example, consider a procedure with two parameters, an integer and a four-character string. Each parameter requires one 32-bit word. Fig. 4-8(a) shows what the parameter portion of a message built by a client stub on an Intel Pentium might look like. The first word contains the integer parameter, 5 in this case, and the second contains the string "JILL."

Figure 4-8. (a) The original message on the Pentium. (b) The message after receipt on the SPARC. (c) The message after being inverted. The little numbers in boxes indicate the address of each byte.



Since messages are transferred byte for byte (actually, bit for bit) over the network, the first byte sent is the first byte to arrive. In Fig. 4-8(b) we show what the message of Fig. 4-8(a) would look like if received by a SPARC, which numbers its bytes with byte 0 at the left (high-order byte) instead of at the right (low-order byte) as do all the Intel chips. When the server stub reads the parameters at addresses 0 and 4, respectively, it will find an integer equal to 83,886,080 (5 x 224) and a string "JILL".

One obvious, but unfortunately incorrect, approach is to simply invert the bytes of each word after they are received, leading to Fig. 4-8(c). Now the integer is 5 and the string is "LLIJ". The problem here is that integers are reversed by the different byte ordering, but strings are not. Without additional information about what is a string and what is an integer, there is no way to repair the damage.
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Passing Reference Parameters

We now come to a difficult problem: How are pointers, or in general, references passed? The answer is: only with the greatest of difficulty, if at all. Remember that a pointer is meaningful only within the address space of the process in which it is being used. Getting back to our read example discussed earlier, if the second parameter (the address of the buffer) happens to be 1000 on the client, one cannot just pass the number 1000 to the server and expect it to work. Address 1000 on the server might be in the middle of the program text.

One solution is just to forbid pointers and reference parameters in general. However, these are so important that this solution is highly undesirable. In fact, it is not necessary either. In the read example, the client stub knows that the second parameter points to an array of characters. Suppose, for the moment, that it also knows how big the array is. One strategy then becomes apparent: copy the array into the message and send it to the server. The server stub can then call the server with a pointer to this array, even though this pointer has a different numerical value than the second parameter of read has. Changes the server makes using the pointer (e.g., storing data into it) directly affect the message buffer inside the server stub. When the server finishes, the original message can be sent back to the client stub, which then copies it back to the client. In effect, call-by-reference has been replaced by copy/restore. Although this is not always identical, it frequently is good enough.

One optimization makes this mechanism twice as efficient. If the stubs know whether the buffer is an input parameter or an output parameter to the server, one of the copies can be eliminated. If the array is input to the server (e.g., in a call to write) it need not be copied back. If it is output, it need not be sent over in the first place.

As a final comment, it is worth noting that although we can now handle pointers to simple arrays and structures, we still cannot handle the most general case of a pointer to an arbitrary data

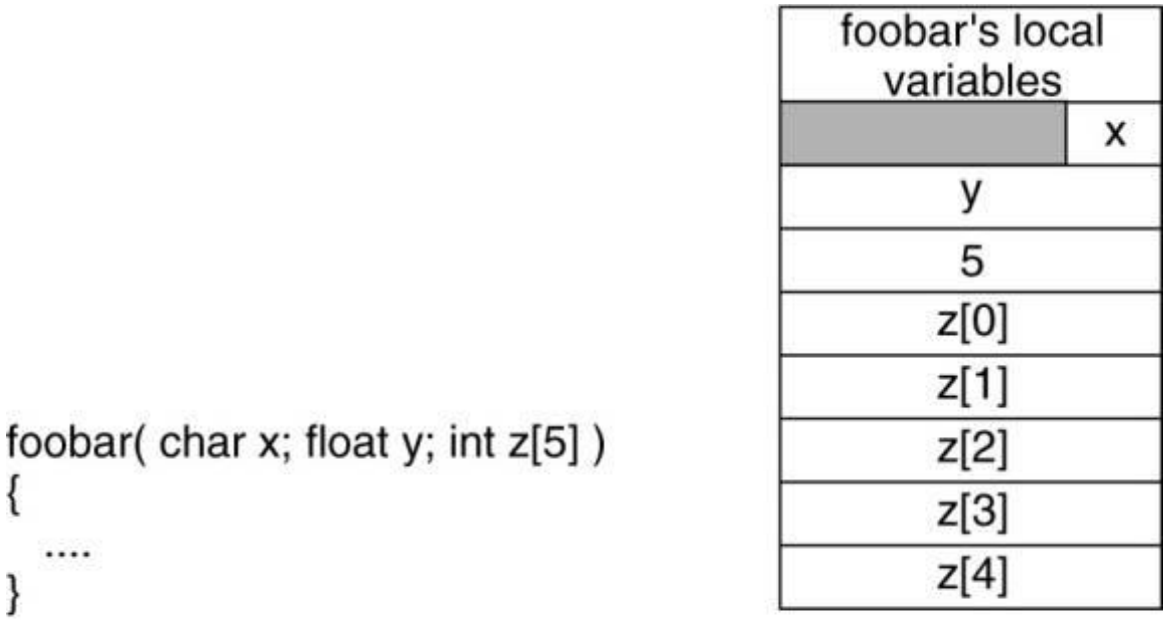
structure such as a complex graph. Some systems attempt to deal with this case by actually passing the pointer to the server stub and generating special code in the server procedure for using pointers. For example, a request may be sent back to the client to provide the referenced data.

Parameter Specification and Stub Generation

From what we have explained so far, it is clear that hiding a remote procedure call requires that the caller and the callee agree on the format of the messages they exchange, and that they follow the same steps when it comes to, for example, passing complex data structures. In other words, both sides in an RPC should follow the same protocol or the RPC will not work correctly. [Page 133]

As a simple example, consider the procedure of Fig. 4-9(a). It has three parameters, a character, a floating-point number, and an array of five integers. Assuming a word is four bytes, the RPC protocol might prescribe that we should transmit a character in the rightmost byte of a word (leaving the next 3 bytes empty), a float as a whole word, and an array as a group of words equal to the array length, preceded by a word giving the length, as shown in Fig. 4-9(b). Thus given these rules, the client stub for foobar knows that it must use the format of Fig. 4-9(b), and the server stub knows that incoming messages for foobar will have the format of Fig. 4-9(b).

Figure 4-9. (a) A procedure. (b) The corresponding message.



Defining the message format is one aspect of an RPC protocol, but it is not sufficient. What we also need is the client and the server to agree on the representation of simple data structures, such as integers, characters, Booleans, etc. For example, the protocol could prescribe that integers are represented in two's complement, characters in 16-bit Unicode, and floats in the IEEE standard #754 format, with everything stored in little endian. With this additional information, messages can be unambiguously interpreted.

With the encoding rules now pinned down to the last bit, the only thing that remains to be done is that the caller and callee agree on the actual exchange of messages. For example, it may be decided to use a connection-oriented transport service such as TCP/IP. An alternative is to use an unreliable datagram service and let the client and server implement an error control scheme as part of the RPC protocol. In practice, several variants exist.

Once the RPC protocol has been fully defined, the client and server stubs need to be implemented. Fortunately, stubs for the same protocol but different procedures normally differ only in their interface to the applications. An interface consists of a collection of procedures that can be called by a client, and which are implemented by a server. An interface is usually available in the same programing language as the one in which the client or server is written (although this is strictly speaking, not necessary). To simplify matters, interfaces are often specified by means of an Interface Definition Language (IDL). An interface specified in such an IDL is then subsequently compiled into a client stub and a server stub, along with the appropriate compile-time or run-time interfaces.
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Practice shows that using an interface definition language considerably simplifies client-server applications based on RPCs. Because it is easy to fully generate client and server stubs, all RPC-based middleware systems offer an IDL to support application development. In some cases, using the IDL is even mandatory, as we shall see in later chapters.

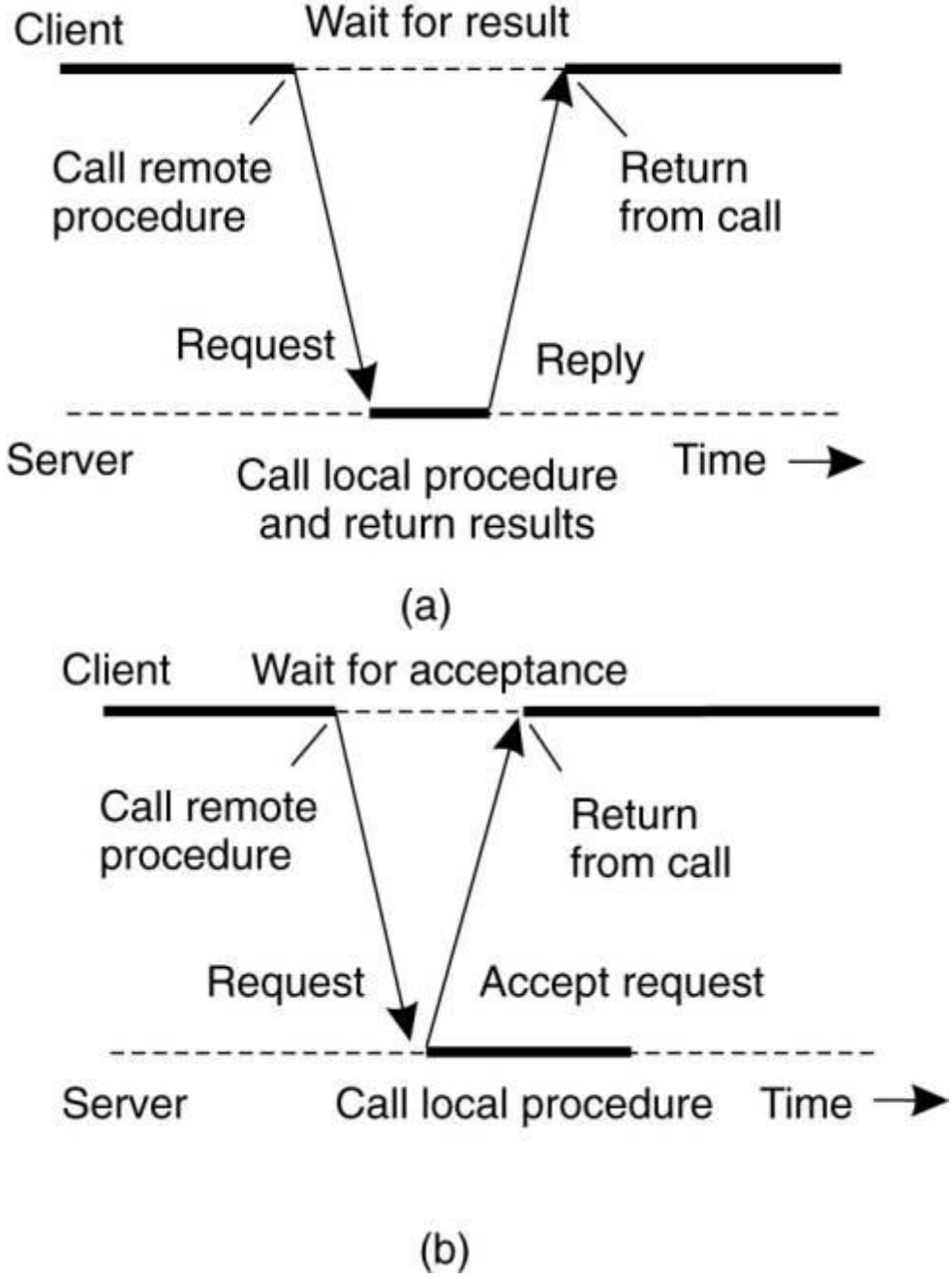
4.2.3. Asynchronous RPC

As in conventional procedure calls, when a client calls a remote procedure, the client will block until a reply is returned. This strict request-reply behavior is unnecessary when there is no result to return, and only leads to blocking the client while it could have proceeded and have done useful work just after requesting the remote procedure to be called. Examples of where there is often no need to wait for a reply include: transferring money from one account to another, adding entries into a database, starting remote services, batch processing, and so on.

To support such situations, RPC systems may provide facilities for what are called asynchronous RPCs, by which a client immediately continues after issuing the RPC request. With asynchronous RPCs, the server immediately sends a reply back to the client the moment

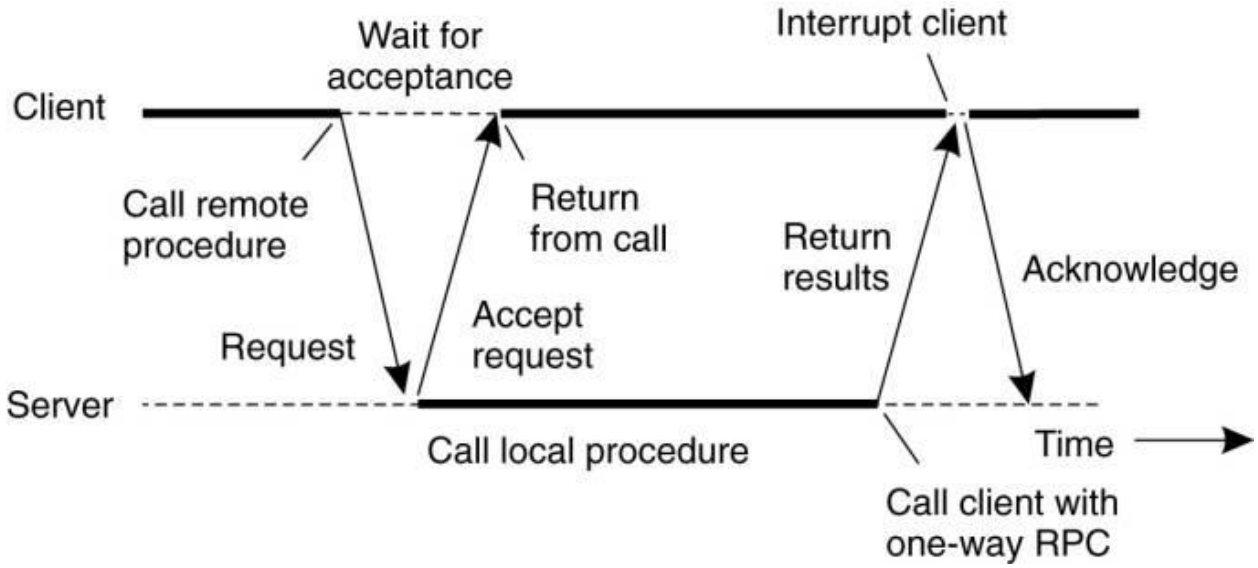
the RPC request is received, after which it calls the requested procedure. The reply acts as an acknowledgment to the client that the server is going to process the RPC. The client will continue without further blocking as soon as it has received the server's acknowledgment. Fig. 4-10(b) shows how client and server interact in the case of asynchronous RPCs. For comparison, Fig. 4-10(a) shows the normal request-reply behavior.

Figure 4-10. (a) The interaction between client and server in a traditional RPC. (b) The interaction using asynchronous RPC.
(This item is displayed on page 135 in the print version)



Asynchronous RPCs can also be useful when a reply will be returned but the client is not prepared to wait for it and do nothing in the meantime. For example, a client may want to prefetch the network addresses of a set of hosts that it expects to contact soon. While a naming service is collecting those addresses, the client may want to do other things. In such cases, it makes sense to organize the communication between the client and server through two asynchronous RPCs, as shown in Fig. 4-11. The client first calls the server to hand over a list of host names that should be looked up, and continues when the server has acknowledged the receipt of that list. The second call is done by the server, who calls the client to hand over the addresses it found. Combining two asynchronous RPCs is sometimes also referred to as a deferred synchronous RPC.

Figure 4-11. A client and server interacting through two asynchronous RPCs.
(This item is displayed on page 135 in the print version)



It should be noted that variants of asynchronous RPCs exist in which the client continues executing immediately after sending the request to the server. In other words, the client does not wait for an acknowledgment of the server's acceptance of the request. We refer to such RPCs as one-way RPCs. The problem with this approach is that when reliability is not guaranteed, the client cannot know for sure whether or not its request will be processed. We return to these matters in Chap. 8. Likewise, in the case of deferred synchronous RPC, the client may poll the server to see whether the results are available yet instead of letting the server calling back the client.
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4.2.4. Example: DCE RPC

Remote procedure calls have been widely adopted as the basis of middleware and distributed systems in general. In this section, we take a closer look at one specific RPC system: the Distributed Computing Environment (DCE), which was developed by the Open Software Foundation (OSF), now called The Open Group. DCE RPC is not as popular as some other RPC systems, notably Sun RPC. However, DCE RPC is nevertheless representative of other RPC systems, and its specifications have been adopted in Microsoft's base system for distributed computing, DCOM (Eddon and Eddon, 1998). We start with a brief introduction to DCE, after which we consider the principal workings of DCE RPC. Detailed technical information on how to develop RPC-based applications can be found in Stevens (1999).
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Introduction to DCE

DCE is a true middleware system in that it is designed to execute as a layer of abstraction between existing (network) operating systems and distributed applications. Initially designed for UNIX, it has now been ported to all major operating systems including VMS and Windows variants, as well as desktop operating systems. The idea is that the customer can take a collection of existing machines, add the DCE software, and then be able to run distributed applications, all without disturbing existing (nondistributed) applications. Although most of the DCE package runs in user space, in some configurations a piece (part of the distributed file system) must be added to the kernel. The Open Group itself only sells source code, which vendors integrate into their systems.

The programming model underlying all of DCE is the client-server model, which was extensively discussed in the previous chapter. User processes act as clients to access remote services provided by server processes. Some of these services are part of DCE itself, but others belong to the applications and are written by the applications programmers. All communication between clients and servers takes place by means of RPCs.

There are a number of services that form part of DCE itself. The distributed file service is a worldwide file system that provides a transparent way of accessing any file in the system in the same way. It can either be built on top of the hosts' native file systems or used instead of them. The directory service is used to keep track of the location of all resources in the system. These resources include machines, printers, servers, data, and much more, and they may be distributed geographically over the entire world. The directory service allows a process to ask for a resource and not have to be concerned about where it is, unless the process cares. The security service allows resources of all kinds to be protected, so access can be restricted to authorized persons. Finally, the distributed time service is a service that attempts to keep clocks on the different machines globally synchronized. As we shall see in later chapters, having some notion of global time makes it much easier to ensure consistency in a distributed system.

Goals of DCE RPC

The goals of the DCE RPC system are relatively traditional. First and foremost, the RPC system makes it possible for a client to access a remote service by simply calling a local procedure. This interface makes it possible for client (i.e., application) programs to be written in a simple way, familiar to most programmers. It also makes it easy to have large volumes of existing code run in a distributed environment with few, if any, changes.
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It is up to the RPC system to hide all the details from the clients, and, to some extent, from the servers as well. To start with, the RPC system can automatically locate the correct server, and subsequently set up the communication between client and server software (generally called binding). It can also handle the message transport in both directions, fragmenting and reassembling them as needed (e.g., if one of the parameters is a large array). Finally, the RPC system can automatically handle data type conversions between the client and the server, even if they run on different architectures and have a different byte ordering.

As a consequence of the RPC system's ability to hide the details, clients and servers are highly independent of one another. A client can be written in Java and a server in C, or vice versa. A client and server can run on different hardware platforms and use different operating systems. A variety of network protocols and data representations are also supported, all without any intervention from the client or server.

Writing a Client and a Server

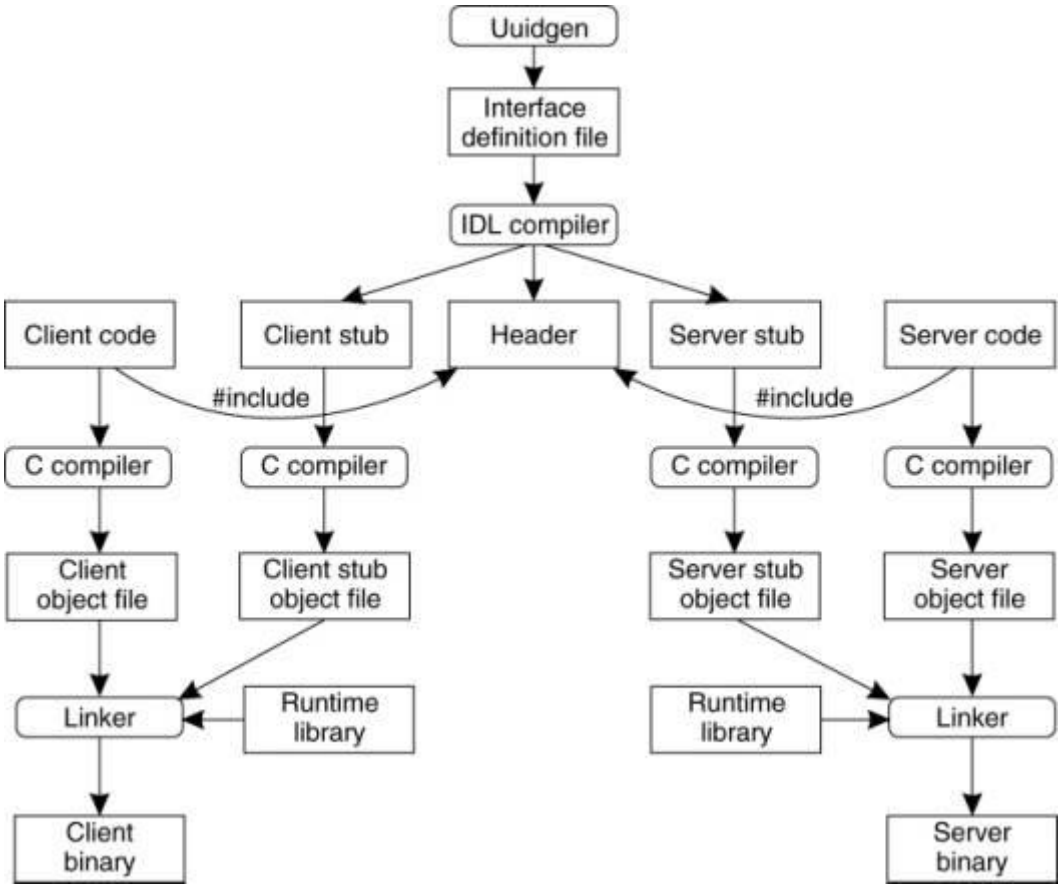
The DCE RPC system consists of a number of components, including languages, libraries, daemons, and utility programs, among others. Together these make it possible to write clients and servers. In this section we will describe the pieces and how they fit together. The entire process of writing and using an RPC client and server is summarized in Fig. 4-12.

In a client-server system, the glue that holds everything together is the interface definition, as specified in the Interface Definition Language, or IDL. It permits procedure declarations in a form closely resembling function prototypes in ANSI C. IDL files can also contain type definitions, constant declarations, and other information needed to correctly marshal parameters and unmarshal results. Ideally, the interface definition should also contain a formal definition of what the procedures do, but such a definition is beyond the current state of the art, so the interface definition just defines the syntax of the calls, not their semantics. At best the writer can add a few comments describing what the procedures do.

A crucial element in every IDL file is a globally unique identifier for the specified interface. The client sends this identifier in the first RPC message and the server verifies that it is correct. In this way, if a client inadvertently tries to bind to the wrong server, or even to an older version of the right server, the server will detect the error and the binding will not take place.

Interface definitions and unique identifiers are closely related in DCE. As illustrated in Fig. 4-12, the first step in writing a client/server application is usually calling the uuidgen program, asking it to generate a prototype IDL file containing an interface identifier guaranteed never to be used again in any interface generated anywhere by uuidgen. Uniqueness is ensured by encoding in it the location and time of creation. It consists of a 128-bit binary number represented in the IDL file as an ASCII string in hexadecimal.
[Page 138]

Figure 4-12. The steps in writing a client and a server in DCE RPC.



The next step is editing the IDL file, filling in the names of the remote procedures and their parameters. It is worth noting that RPC is not totally transparent—for example, the client and

server cannot share global variables—but the IDL rules make it impossible to express constructs that are not supported.

When the IDL file is complete, the IDL compiler is called to process it. The output of the IDL compiler consists of three files:

1. A header file (e.g., interface.h, in C terms).

2. The client stub.

3. The server stub.

The header file contains the unique identifier, type definitions, constant definitions, and function prototypes. It should be included (using #include) in both the client and server code. The client stub contains the actual procedures that the client program will call. These procedures are the ones responsible for collecting and packing the parameters into the outgoing message and then calling the runtime system to send it. The client stub also handles unpacking the reply and returning values to the client. The server stub contains the procedures called by the runtime system on the server machine when an incoming message arrives. These, in turn, call the actual server procedures that do the work.
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The next step is for the application writer to write the client and server code. Both of these are then compiled, as are the two stub procedures. The resulting client code and client stub object files are then linked with the runtime library to produce the executable binary for the client. Similarly, the server code and server stub are compiled and linked to produce the server's binary. At runtime, the client and server are started so that the application is actually executed as well.

Binding a Client to a Server

To allow a client to call a server, it is necessary that the server be registered and prepared to accept incoming calls. Registration of a server makes it possible for a client to locate the server and bind to it. Server location is done in two steps:

1. Locate the server's machine.

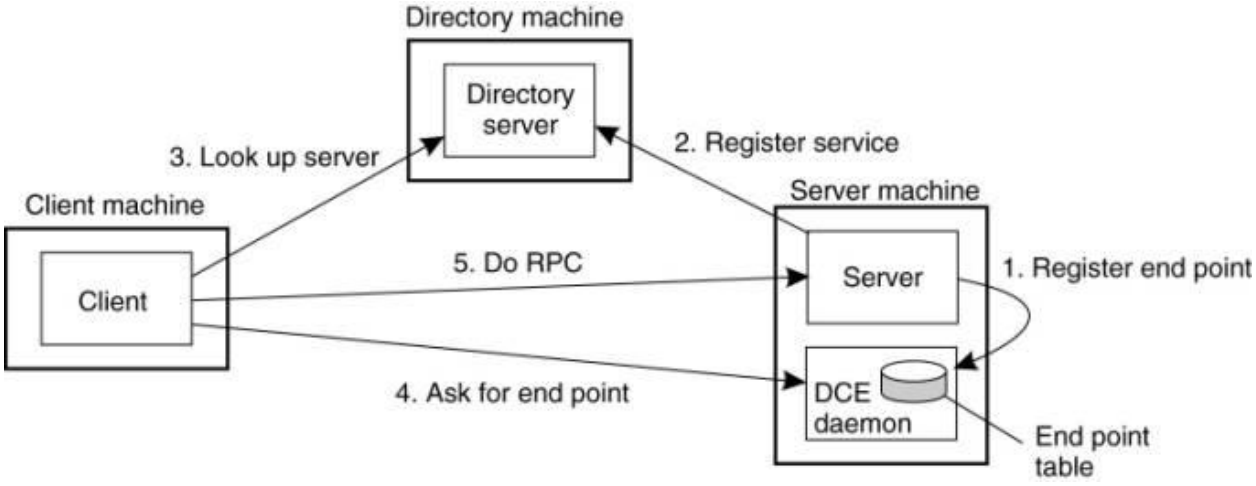
2. Locate the server (i.e., the correct process) on that machine.

The second step is somewhat subtle. Basically, what it comes down to is that to communicate with a server, the client needs to know an end point, on the server's machine to which it can send messages. An end point (also commonly known as a port) is used by the server's

operating system to distinguish incoming messages for different processes. In DCE, a table of (server, end point) pairs is maintained on each server machine by a process called the DCE daemon. Before it becomes available for incoming requests, the server must ask the operating system for an end point. It then registers this end point with the DCE daemon. The DCE daemon records this information (including which protocols the server speaks) in the end point table for future use.

The server also registers with the directory service by providing it the network address of the server's machine and a name under which the server can be looked up. Binding a client to a server then proceeds as shown in Fig. 4-13.

Figure 4-13. Client-to-server binding in DCE.
(This item is displayed on page 140 in the print version)



Let us assume that the client wants to bind to a video server that is locally known under the name /local/multimedia/video/movies. It passes this name to the directory server, which returns the network address of the machine running the video server. The client then goes to the DCE daemon on that machine (which has a well-known end point), and asks it to look up the end point of the video server in its end point table. Armed with this information, the RPC can now take place. On subsequent RPCs this lookup is not needed. DCE also gives clients the ability to do more sophisticated searches for a suitable server when that is needed. Secure RPC is also an option where confidentiality or data integrity is crucial.
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Performing an RPC

The actual RPC is carried out transparently and in the usual way. The client stub marshals the parameters to the runtime library for transmission using the protocol chosen at binding time.

When a message arrives at the server side, it is routed to the correct server based on the end point contained in the incoming message. The runtime library passes the message to the server stub, which unmarshals the parameters and calls the server. The reply goes back by the reverse route.

DCE provides several semantic options. The default is at-most-once operation, in which case no call is ever carried out more than once, even in the face of system crashes. In practice, what this means is that if a server crashes during an RPC and then recovers quickly, the client does not repeat the operation, for fear that it might already have been carried out once.

Alternatively, it is possible to mark a remote procedure as idempotent (in the IDL file), in which case it can be repeated multiple times without harm. For example, reading a specified block from a file can be tried over and over until it succeeds. When an idempotent RPC fails due to a server crash, the client can wait until the server reboots and then try again. Other semantics are also available (but rarely used), including broadcasting the RPC to all the machines on the local network. We return to RPC semantics in Chap. 8, when discussing RPC in the presence of failures.

4.3. Message-Oriented Communication

Remote procedure calls and remote object invocations contribute to hiding communication in distributed systems, that is, they enhance access transparency. Unfortunately, neither mechanism is always appropriate. In particular, when it cannot be assumed that the receiving side is executing at the time a request is issued, alternative communication services are needed. Likewise, the inherent synchronous nature of RPCs, by which a client is blocked until its request has been processed, sometimes needs to be replaced by something else.
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That something else is messaging. In this section we concentrate on message-oriented communication in distributed systems by first taking a closer look at what exactly synchronous behavior is and what its implications are. Then, we discuss messaging systems that assume that parties are executing at the time of communication. Finally, we will examine message-queuing systems that allow processes to exchange information, even if the other party is not executing at the time communication is initiated.

4.3.1. Message-Oriented Transient Communication

Many distributed systems and applications are built directly on top of the simple message-oriented model offered by the transport layer. To better understand and appreciate the message-oriented systems as part of middleware solutions, we first discuss messaging through transport-level sockets.

Berkeley Sockets

Special attention has been paid to standardizing the interface of the transport layer to allow programmers to make use of its entire suite of (messaging) protocols through a simple set of primitives. Also, standard interfaces make it easier to port an application to a different machine.

As an example, we briefly discuss the sockets interface as introduced in the 1970s in Berkeley UNIX. Another important interface is XTI, which stands for the X/Open Transport Interface, formerly called the Transport Layer Interface (TLI), and developed by AT&T. Sockets and XTI are very similar in their model of network programming, but differ in their set of primitives.

Conceptually, a socket is a communication end point to which an application can write data that are to be sent out over the underlying network, and from which incoming data can be read. A socket forms an abstraction over the actual communication end point that is used by the local operating system for a specific transport protocol. In the following text, we concentrate on the socket primitives for TCP, which are shown in Fig. 4-14.

Figure 4-14. The socket primitives for TCP/IP.
(This item is displayed on page 142 in the print version)

Primitive	Meaning
Socket	Create a new communication end point
Bind	Attach a local address to a socket
Listen	Announce willingness to accept connections
Accept	Block caller until a connection request arrives
Connect	Actively attempt to establish a connection
Send	Send some data over the connection
Receive	Receive some data over the connection
Close	Release the connection

Servers generally execute the first four primitives, normally in the order given. When calling the socket primitive, the caller creates a new communication end point for a specific transport protocol. Internally, creating a communication end point means that the local operating system reserves resources to accommodate sending and receiving messages for the specified protocol.

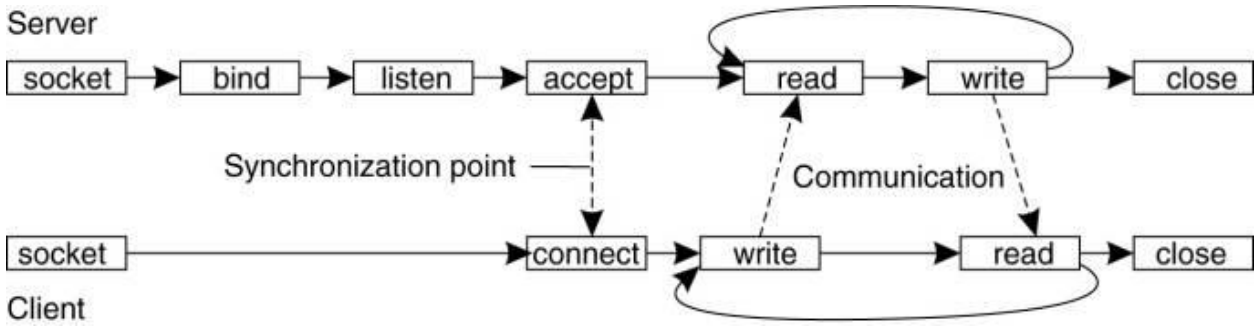
The bind primitive associates a local address with the newly-created socket. For example, a server should bind the IP address of its machine together with a (possibly well-known) port number to a socket. Binding tells the operating system that the server wants to receive messages only on the specified address and port.
[Page 142]

The listen primitive is called only in the case of connection-oriented communication. It is a nonblocking call that allows the local operating system to reserve enough buffers for a specified maximum number of connections that the caller is willing to accept.

A call to accept blocks the caller until a connection request arrives. When a request arrives, the local operating system creates a new socket with the same properties as the original one, and returns it to the caller. This approach will allow the server to, for example, fork off a process that will subsequently handle the actual communication through the new connection. The server, in the meantime, can go back and wait for another connection request on the original socket.

Let us now take a look at the client side. Here, too, a socket must first be created using the socket primitive, but explicitly binding the socket to a local address is not necessary, since the operating system can dynamically allocate a port when the connection is set up. The connect primitive requires that the caller specifies the transport-level address to which a connection request is to be sent. The client is blocked until a connection has been set up successfully, after which both sides can start exchanging information through the send and receive primitives. Finally, closing a connection is symmetric when using sockets, and is established by having both the client and server call the close primitive. The general pattern followed by a client and server for connection-oriented communication using sockets is shown in Fig. 4-15. Details about network programming using sockets and other interfaces in a UNIX environment can be found in Stevens (1998).

Figure 4-15. Connection-oriented communication pattern using sockets.
(This item is displayed on page 143 in the print version)



The Message-Passing Interface (MPI)

With the advent of high-performance multicomputers, developers have been looking for message-oriented primitives that would allow them to easily write highly efficient applications. This means that the primitives should be at a convenient level of abstraction (to ease application development), and that their implementation incurs only minimal overhead. Sockets were deemed insufficient for two reasons. First, they were at the wrong level of abstraction by supporting only simple send and receive primitives. Second, sockets had been designed to communicate across networks using general-purpose protocol stacks such as TCP/IP. They were not considered suitable for the proprietary protocols developed for high-speed interconnection networks, such as those used in high-performance server clusters. Those protocols required an interface that could handle more advanced features, such as different forms of buffering and synchronization.
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The result was that most interconnection networks and high-performance multicomputers were shipped with proprietary communication libraries. These libraries offered a wealth of high-level and generally efficient communication primitives. Of course, all libraries were mutually incompatible, so that application developers now had a portability problem.

The need to be hardware and platform independent eventually led to the definition of a standard for message passing, simply called the Message-Passing Interface or MPI. MPI is designed for parallel applications and as such is tailored to transient communication. It makes direct use of the underlying network. Also, it assumes that serious failures such as process crashes or network partitions are fatal and do not require automatic recovery.

MPI assumes communication takes place within a known group of processes. Each group is assigned an identifier. Each process within a group is also assigned a (local) identifier. A (groupID, processID) pair therefore uniquely identifies the source or destination of a message, and is used instead of a transport-level address. There may be several, possibly overlapping groups of processes involved in a computation and that are all executing at the same time.

At the core of MPI are messaging primitives to support transient communication, of which the most intuitive ones are summarized in Fig. 4-16.

Figure 4-16. Some of the most intuitive message-passing primitives of MPI.
(This item is displayed on page 144 in the print version)

Primitive	Meaning
MPI_bsend	Append outgoing message to a local send buffer
MPI_send	Send a message and wait until copied to local or remote buffer
MPI_ssend	Send a message and wait until receipt starts

MPI_sendrecv	Send a message and wait for reply
MPI_isend	Pass reference to outgoing message, and continue
MPI_issend	Pass reference to outgoing message, and wait until receipt starts
MPI_recv	Receive a message; block if there is none
MPI_irecv	Check if there is an incoming message, but do not block

Transient asynchronous communication is supported by means of the MPI_bsend primitive. The sender submits a message for transmission, which is generally first copied to a local buffer in the MPI runtime system. When the message has been copied, the sender continues. The local MPI runtime system will remove the message from its local buffer and take care of transmission as soon as a receiver has called a receive primitive.
[Page 144]

There is also a blocking send operation, called MPI_send, of which the semantics are implementation dependent. The primitive MPI_send may either block the caller until the specified message has been copied to the MPI runtime system at the sender's side, or until the receiver has initiated a receive operation. Synchronous communication by which the sender blocks until its request is accepted for further processing is available through the MPI_ssend primitive. Finally, the strongest form of synchronous communication is also supported: when a sender calls MPI_sendrecv, it sends a request to the receiver and blocks until the latter returns a reply. Basically, this primitive corresponds to a normal RPC.

Both MPI_send and MPI_ssend have variants that avoid copying messages from user buffers to buffers internal to the local MPI runtime system. These variants correspond to a form of asynchronous communication. With MPI_isend, a sender passes a pointer to the message after which the MPI runtime system takes care of communication. The sender immediately continues. To prevent overwriting the message before communication completes, MPI offers primitives to check for completion, or even to block if required. As with MPI_send, whether the message has actually been transferred to the receiver or that it has merely been copied by the local MPI runtime system to an internal buffer is left unspecified.

Likewise, with MPI_issend, a sender also passes only a pointer to the MPI runtime system. When the runtime system indicates it has processed the message, the sender is then guaranteed that the receiver has accepted the message and is now working on it.

The operation MPI_recv is called to receive a message; it blocks the caller until a message arrives. There is also an asynchronous variant, called MPI_irecv, by which a receiver indicates

that is prepared to accept a message. The receiver can check whether or not a message has indeed arrived, or block until one does.

The semantics of MPI communication primitives are not always straightforward, and different primitives can sometimes be interchanged without affecting the correctness of a program. The official reason why so many different forms of communication are supported is that it gives implementers of MPI systems enough possibilities for optimizing performance. Cynics might say the committee could not make up its collective mind, so it threw in everything. MPI has been designed for high-performance parallel applications, which makes it easier to understand its diversity in different communication primitives.
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More on MPI can be found in Gropp et al. (1998b) The complete reference in which the over 100 functions in MPI are explained in detail, can be found in Snir et al. (1998) and Gropp et al. (1998a)

4.3.2. Message-Oriented Persistent Communication

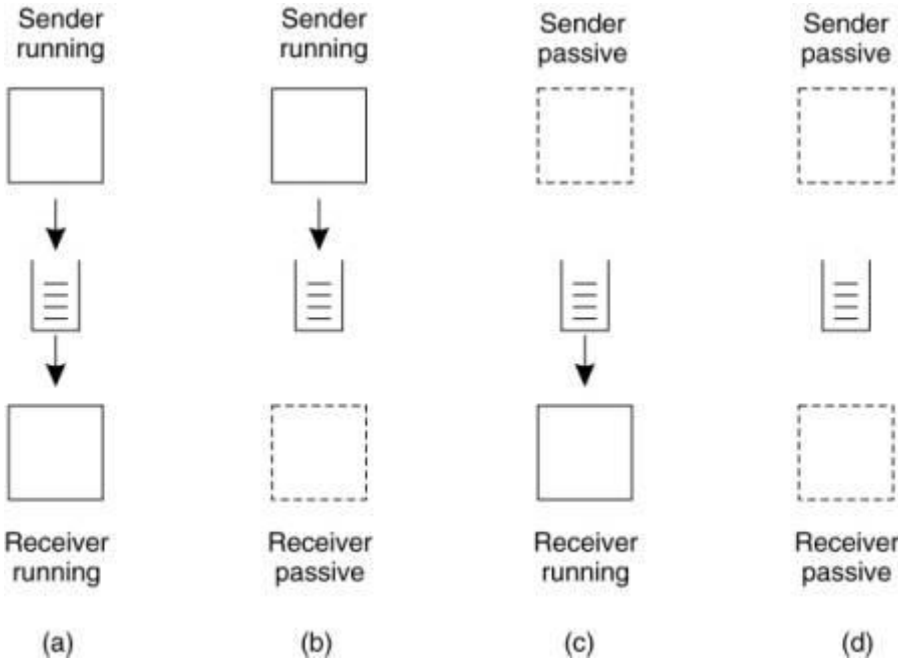
We now come to an important class of message-oriented middleware services, generally known as message-queuing systems, or just Message-Oriented Middleware (MOM). Message-queuing systems provide extensive support for persistent asynchronous communication. The essence of these systems is that they offer intermediate-term storage capacity for messages, without requiring either the sender or receiver to be active during message transmission. An important difference with Berkeley sockets and MPI is that message-queuing systems are typically targeted to support message transfers that are allowed to take minutes instead of seconds or milliseconds. We first explain a general approach to message-queuing systems, and conclude this section by comparing them to more traditional systems, notably the Internet e-mail systems.
Message-Queuing Model

The basic idea behind a message-queuing system is that applications communicate by inserting messages in specific queues. These messages are forwarded over a series of communication servers and are eventually delivered to the destination, even if it was down when the message was sent. In practice, most communication servers are directly connected to each other. In other words, a message is generally transferred directly to a destination server. In principle, each application has its own private queue to which other applications can send messages. A queue can be read only by its associated application, but it is also possible for multiple applications to share a single queue.

An important aspect of message-queuing systems is that a sender is generally given only the guarantees that its message will eventually be inserted in the recipient's queue. No guarantees are given about when, or even if the message will actually be read, which is completely determined by the behavior of the recipient.

These semantics permit communication loosely-coupled in time. There is thus no need for the receiver to be executing when a message is being sent to its queue. Likewise, there is no need for the sender to be executing at the moment its message is picked up by the receiver. The sender and receiver can execute completely independently of each other. In fact, once a message has been deposited in a queue, it will remain there until it is removed, irrespective of whether its sender or receiver is executing. This gives us four combinations with respect to the execution mode of the sender and receiver, as shown in Fig. 4-17.
[Page 146]

Figure 4-17. Four combinations for loosely-coupled communications using queues.



In Fig. 4-17(a), both the sender and receiver execute during the entire transmission of a message. In Fig. 4-17(b), only the sender is executing, while the receiver is passive, that is, in a state in which message delivery is not possible. Nevertheless, the sender can still send messages. The combination of a passive sender and an executing receiver is shown in Fig. 4-17(c). In this case, the receiver can read messages that were sent to it, but it is not necessary that their respective senders are executing as well. Finally, in Fig. 4-17(d), we see the situation that the system is storing (and possibly transmitting) messages even while sender and receiver are passive.

Messages can, in principle, contain any data. The only important aspect from the perspective of middleware is that messages are properly addressed. In practice, addressing is done by providing a systemwide unique name of the destination queue. In some cases, message size may be limited, although it is also possible that the underlying system takes care of fragmenting

and assembling large messages in a way that is completely transparent to applications. An effect of this approach is that the basic interface offered to applications can be extremely simple, as shown in Fig. 4-18.

Figure 4-18. Basic interface to a queue in a message-queuing system.
(This item is displayed on page 147 in the print version)

Primitive	Meaning
Put	Append a message to a specified queue
Get	Block until the specified queue is nonempty, and remove the first message
Poll	Check a specified queue for messages, and remove the first. Never block
Notify	Install a handler to be called when a message is put into the specified queue

The put primitive is called by a sender to pass a message to the underlying system that is to be appended to the specified queue. As we explained, this is a nonblocking call. The get primitive is a blocking call by which an authorized process can remove the longest pending message in the specified queue. The process is blocked only if the queue is empty. Variations on this call allow searching for a specific message in the queue, for example, using a priority, or a matching pattern. The nonblocking variant is given by the poll primitive. If the queue is empty, or if a specific message could not be found, the calling process simply continues.
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Finally, most queuing systems also allow a process to install a handler as a callback function, which is automatically invoked whenever a message is put into the queue. Callbacks can also be used to automatically start a process that will fetch messages from the queue if no process is currently executing. This approach is often implemented by means of a daemon on the receiver's side that continuously monitors the queue for incoming messages and handles accordingly.

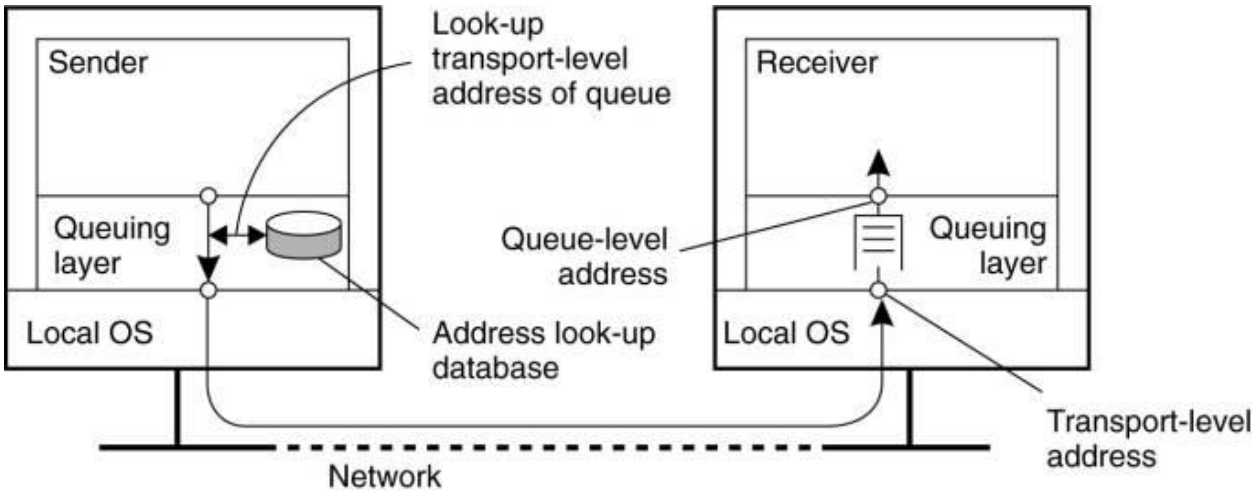
General Architecture of a Message-Queuing System

Let us now take a closer look at what a general message-queuing system looks like. One of the first restrictions that we make is that messages can be put only into queues that are local to the sender, that is, queues on the same machine, or no worse than on a machine nearby such as on the same LAN that can be efficiently reached through an RPC. Such a queue is called the

source queue. Likewise, messages can be read only from local queues. However, a message put into a queue will contain the specification of a destination queue to which it should be transferred. It is the responsibility of a message-queuing system to provide queues to senders and receivers and take care that messages are transferred from their source to their destination queue.

It is important to realize that the collection of queues is distributed across multiple machines. Consequently, for a message-queuing system to transfer messages, it should maintain a mapping of queues to network locations. In practice, this means that it should maintain a (possibly distributed) database of queue names to network locations, as shown in Fig. 4-19. Note that such a mapping is completely analogous to the use of the Domain Name System (DNS) for e-mail in the Internet. For example, when sending a message to the logical mail address steen@cs.vu.nl, the mailing system will query DNS to find the network (i.e., IP) address of the recipient's mail server to use for the actual message transfer.
[Page 148]

Figure 4-19. The relationship between queue-level addressing and network-level addressing.

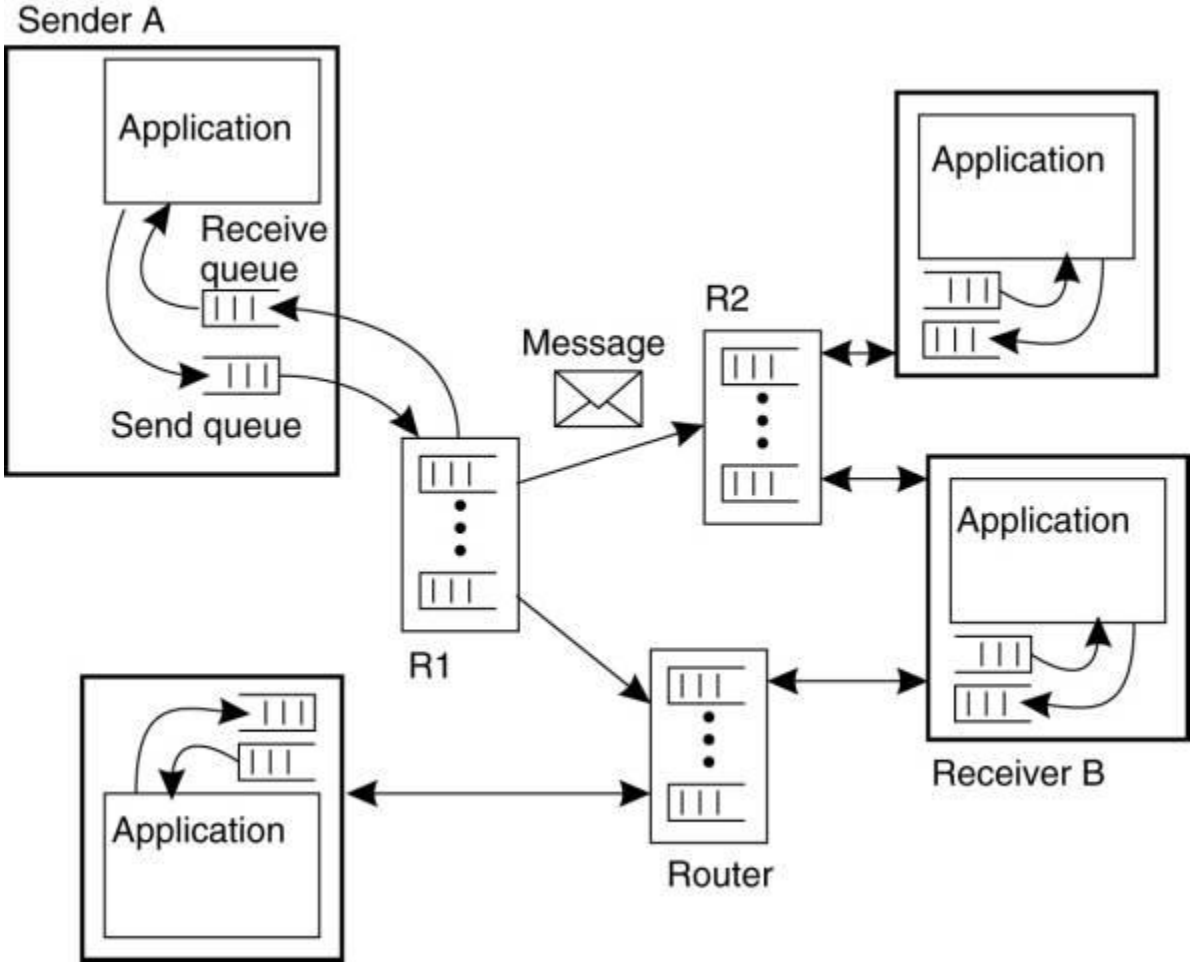


Queues are managed by queue managers. Normally, a queue manager interacts directly with the application that is sending or receiving a message. However, there are also special queue managers that operate as routers, or relays: they forward incoming messages to other queue managers. In this way, a message-queuing system may gradually grow into a complete, application-level, overlay network, on top of an existing computer network. This approach is similar to the construction of the early MBone over the Internet, in which ordinary user processes were configured as multicast routers. As it turns out, multicasting through overlay networks is still important as we will discuss later in this chapter.

Relays can be convenient for a number of reasons. For example, in many message-queuing systems, there is no general naming service available that can dynamically maintain queue-to-location mappings. Instead, the topology of the queuing network is static, and each queue manager needs a copy of the queue-to location mapping. It is needless to say that in large-scale queuing systems, this approach can easily lead to network-management problems.

One solution is to use a few routers that know about the network topology. When a sender A puts a message for destination B in its local queue, that message is first transferred to the nearest router, say R1, as shown in Fig. 4-20. At that point, the router knows what to do with the message and forwards it in the direction of B. For example, R1 may derive from B's name that the message should be forwarded to router R2. In this way, only the routers need to be updated when queues are added or removed, while every other queue manager has to know only where the nearest router is.

Figure 4-20. The general organization of a message-queuing system with routers.
(This item is displayed on page 149 in the print version)



Relays can thus generally help build scalable message-queuing systems. However, as queuing networks grow, it is clear that the manual configuration of networks will rapidly become completely unmanageable. The only solution is to adopt dynamic routing schemes as is done for computer networks. In that respect, it is somewhat surprising that such solutions are not yet integrated into some of the popular message-queuing systems.
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Another reason why relays are used is that they allow for secondary processing of messages. For example, messages may need to be logged for reasons of security or fault tolerance. A special form of relay that we discuss in the next section is one that acts as a gateway, transforming messages into a format that can be understood by the receiver.

Finally, relays can be used for multicasting purposes. In that case, an incoming message is simply put into each send queue.

Message Brokers

An important application area of message-queuing systems is integrating existing and new applications into a single, coherent distributed information system. Integration requires that applications can understand the messages they receive. In practice, this requires the sender to have its outgoing messages in the same format as that of the receiver.

The problem with this approach is that each time an application is added to the system that requires a separate message format, each potential receiver will have to be adjusted in order to produce that format.

An alternative is to agree on a common message format, as is done with traditional network protocols. Unfortunately, this approach will generally not work for message-queuing systems. The problem is the level of abstraction at which these systems operate. A common message format makes sense only if the collection of processes that make use of that format indeed have enough in common. If the collection of applications that make up a distributed information system is highly diverse (which it often is), then the best common format may well be no more than a sequence of bytes.
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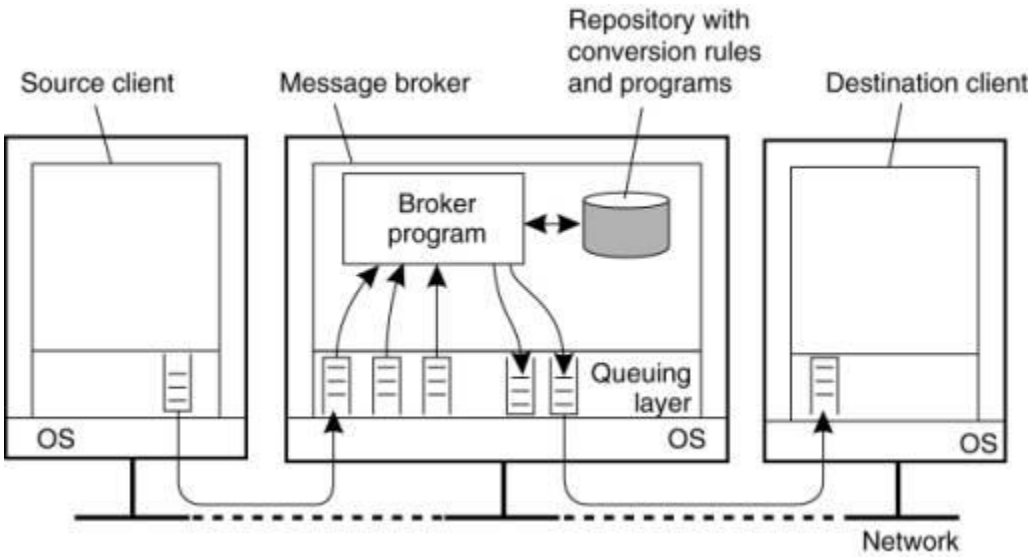
Although a few common message formats for specific application domains have been defined, the general approach is to learn to live with different formats, and try to provide the means to make conversions as simple as possible. In message-queuing systems, conversions are handled by special nodes in a queuing network, known as message brokers. A message broker acts as an application-level gateway in a message-queuing system. Its main purpose is to convert incoming messages so that they can be understood by the destination application. Note that to a message-queuing system, a message broker is just another application, as shown in Fig. 4-21. In other words, a message broker is generally not considered to be an integral part of the queuing system.

Figure 4-21. The general organization of a message broker in a message-queuing system.



A message broker can be as simple as a reformatter for messages. For example, assume an incoming message contains a table from a database, in which records are separated by a special end-of-record delimiter and fields within a record have a known, fixed length. If the destination application expects a different delimiter between records, and also expects that fields have variable lengths, a message broker can be used to convert messages to the format expected by the destination.

In a more advanced setting, a message broker may act as an application-level gateway, such as one that handles the conversion between two different database applications. In such cases, frequently it cannot be guaranteed that all information contained in the incoming message can actually be transformed into something appropriate for the outgoing message.
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However, more common is the use of a message broker for advanced enterprise application integration (EAI) as we discussed in Chap. 1. In this case, rather than (only) converting messages, a broker is responsible for matching applications based on the messages that are being exchanged. In such a model, called publish/subscribe, applications send messages in the form of publishing. In particular, they may publish a message on topic X, which is then sent to the broker. Applications that have stated their interest in messages on topic X, that is, who have subscribed to those messages, will then receive these messages from the broker. More advanced forms of mediation are also possible, but we will defer further discussion until Chap. 13.

At the heart of a message broker lies a repository of rules and programs that can transform a message of type T1 to one of type T2. The problem is defining the rules and developing the programs. Most message broker products come with sophisticated development tools, but the bottom line is still that the repository needs to be filled by experts. Here we see a perfect example where commercial products are often misleadingly said to provide "intelligence," where, in fact, the only intelligence is to be found in the heads of those experts.

A Note on Message-Queuing Systems

Considering what we have said about message-queuing systems, it would appear that they have long existed in the form of implementations for e-mail services. E-mail systems are generally implemented through a collection of mail servers that store and forward messages on behalf of the users on hosts directly connected to the server. Routing is generally left out, as e-mail systems can make direct use of the underlying transport services. For example, in the mail protocol for the Internet, SMTP (Postel, 1982), a message is transferred by setting up a direct TCP connection to the destination mail server.

What makes e-mail systems special compared to message-queuing systems is that they are primarily aimed at providing direct support for end users. This explains, for example, why a number of groupware applications are based directly on an e-mail system (Khoshafian and Buckiewicz 1995). In addition, e-mail systems may have very specific requirements such as automatic message filtering, support for advanced messaging databases (e.g., to easily retrieve previously stored messages), and so on.

General message-queuing systems are not aimed at supporting only end users. An important issue is that they are set up to enable persistent communication between processes, regardless of whether a process is running a user application, handling access to a database, performing computations, and so on. This approach leads to a different set of requirements for message-queuing systems than pure email systems. For example, e-mail systems generally need not provide guaranteed message delivery, message priorities, logging facilities, efficient multicasting, load balancing, fault tolerance, and so on for general usage.
[Page 152]

General-purpose message-queuing systems, therefore, have a wide range of applications, including e-mail, workflow, groupware, and batch processing. However, as we have stated before, the most important application area is the integration of a (possibly widely-dispersed) collection of databases and applications into a federated information system (Hohpe and Woolf, 2004). For example, a query expanding several databases may need to be split into subqueries that are forwarded to individual databases. Message-queuing systems assist by providing the basic means to package each subquery into a message and routing it to the appropriate database. Other communication facilities we have discussed in this chapter are far less appropriate.

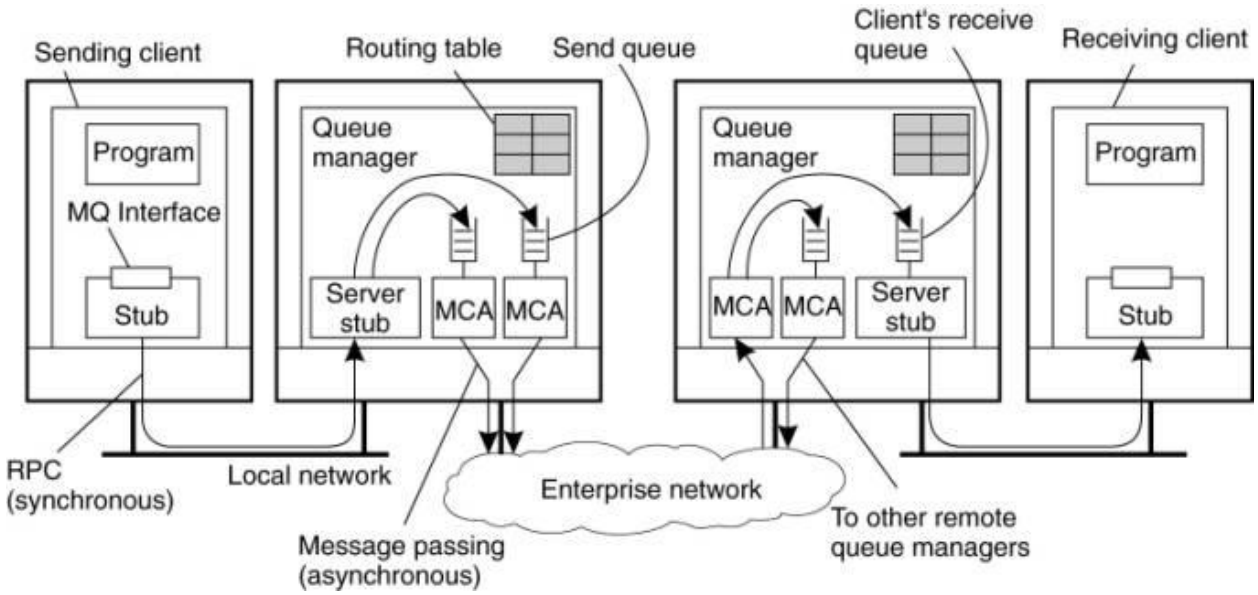
4.3.3. Example: IBM's WebSphere Message-Queuing System

To help understand how message-queuing systems work in practice, let us take a look at one specific system, namely the message-queuing system that is part of IBM's WebSphere product. Formerly known as MQSeries, it is now referred to as WebSphere MQ. There is a wealth of documentation on Web-Sphere MQ, and in the following we can only resort to the basic principles. Many architectural details concerning message-queuing networks can be found in IBM (2005b, 2005d). Programming message-queuing networks is not something that can be learned on a Sunday afternoon, and MQ's programming guide (IBM, 2005a) is a good example showing that going from principles to practice may require substantial effort.

Overview

The basic architecture of an MQ queuing network is quite straightforward, and is shown in Fig. 4-22. All queues are managed by queue managers. A queue manager is responsible for removing messages from its send queues, and forwarding those to other queue managers. Likewise, a queue manager is responsible for handling incoming messages by picking them up from the underlying network and subsequently storing each message in the appropriate input queue. To give an impression of what messaging can mean: a message has a maximum default size of 4 MB, but this can be increased up to 100 MB. A queue is normally restricted to 2 GB of data, but depending on the underlying operating system, this maximum can be easily set higher.

Figure 4-22. General organization of IBM's message-queuing system.
(This item is displayed on page 153 in the print version)



Queue managers are pairwise connected through message channels, which are an abstraction of transport-level connections. A message channel is a unidirectional, reliable connection between a sending and a receiving queue manager, through which queued messages are transported. For example, an Internet-based message channel is implemented as a TCP connection. Each of the two ends of a message channel is managed by a message channel agent (MCA). A sending MCA is basically doing nothing else than checking send queues for a message, wrapping it into a transport-level packet, and sending it along the connection to its associated receiving MCA. Likewise, the basic task of a receiving MCA is listening for an incoming packet, unwrapping it, and subsequently storing the unwrapped message into the appropriate queue.
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Queue managers can be linked into the same process as the application for which it manages the queues. In that case, the queues are hidden from the application behind a standard interface, but effectively can be directly manipulated by the application. An alternative organization is one in which queue managers and applications run on separate machines. In that case, the application is offered the same interface as when the queue manager is colocated on the same machine. However, the interface is implemented as a proxy that communicates with the queue manager using traditional RPC-based synchronous communication. In this way, MQ basically retains the model that only queues local to an application can be accessed.
Channels

An important component of MQ is formed by the message channels. Each message channel has exactly one associated send queue from which it fetches the messages it should transfer to the other end. Transfer along the channel can take place only if both its sending and receiving MCA are up and running. Apart from starting both MCAs manually, there are several alternative ways to start a channel, some of which we discuss next.
[Page 154]

One alternative is to have an application directly start its end of a channel by activating the sending or receiving MCA. However, from a transparency point of view, this is not a very attractive alternative. A better approach to start a sending MCA is to configure the channel's send queue to set off a trigger when a message is first put into the queue. That trigger is associated with a handler to start the sending MCA so that it can remove messages from the send queue.

Another alternative is to start an MCA over the network. In particular, if one side of a channel is already active, it can send a control message requesting that the other MCA to be started. Such a control message is sent to a daemon listening to a well-known address on the same machine as where the other MCA is to be started.

Channels are stopped automatically after a specified time has expired during which no more messages were dropped into the send queue.

Each MCA has a set of associated attributes that determine the overall behavior of a channel. Some of the attributes are listed in Fig. 4-23. Attribute values of the sending and receiving MCA should be compatible and perhaps negotiated first before a channel can be set up. For example, both MCAs should obviously support the same transport protocol. An example of a nonnegotiable attribute is whether or not messages are to be delivered in the same order as they are put into the send queue. If one MCA wants FIFO delivery, the other must comply. An example of a negotiable attribute value is the maximum message length, which will simply be chosen as the minimum value specified by either MCA.

Figure 4-23. Some attributes associated with message channel agents.

Attribute	Description
Transport type	Determines the transport protocol to be used
FIFO delivery	Indicates that messages are to be delivered in the order they are sent
Message length	Maximum length of a single message
Setup retry count	Specifies maximum number of retries to start up the remote MCA
Delivery retries	Maximum times MCA will try to put received message into queue

Message Transfer

To transfer a message from one queue manager to another (possibly remote) queue manager, it is necessary that each message carries its destination address, for which a transmission header is used. An address in MQ consists of two parts. The first part consists of the name of the queue manager to which the message is to be delivered. The second part is the name of the destination queue resorting under that manager to which the message is to be appended.

Besides the destination address, it is also necessary to specify the route that a message should follow. Route specification is done by providing the name of the local send queue to which a message is to be appended. Thus it is not necessary to provide the full route in a message. Recall that each message channel has exactly one send queue. By telling to which send queue

a message is to be appended, we efectively specify to which queue manager a message is to be forwarded.
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In most cases, routes are explicitly stored inside a queue manager in a routing table. An entry in a routing table is a pair (destQM, sendQ), where destQM is the name of the destination queue manager, and sendQ is the name of the local send queue to which a message for that queue manager should be appended. (A routing table entry is called an alias in MQ.)

It is possible that a message needs to be transferred across multiple queue managers before reaching its destination. Whenever such an intermediate queue manager receives the message, it simply extracts the name of the destination queue manager from the message header, and does a routing-table look-up to find the local send queue to which the message should be appended.

It is important to realize that each queue manager has a systemwide unique name that is effectively used as an identifier for that queue manager. The problem with using these names is that replacing a queue manager, or changing its name, will affect all applications that send messages to it. Problems can be alleviated by using a local alias for queue manager names. An alias defined within a queue manager M1 is another name for a queue manager M2, but which is available only to applications interfacing to M1. An alias allows the use of the same (logical) name for a queue, even if the queue manager of that queue changes. Changing the name of a queue manager requires that we change its alias in all queue managers. However, applications can be left unaffected.
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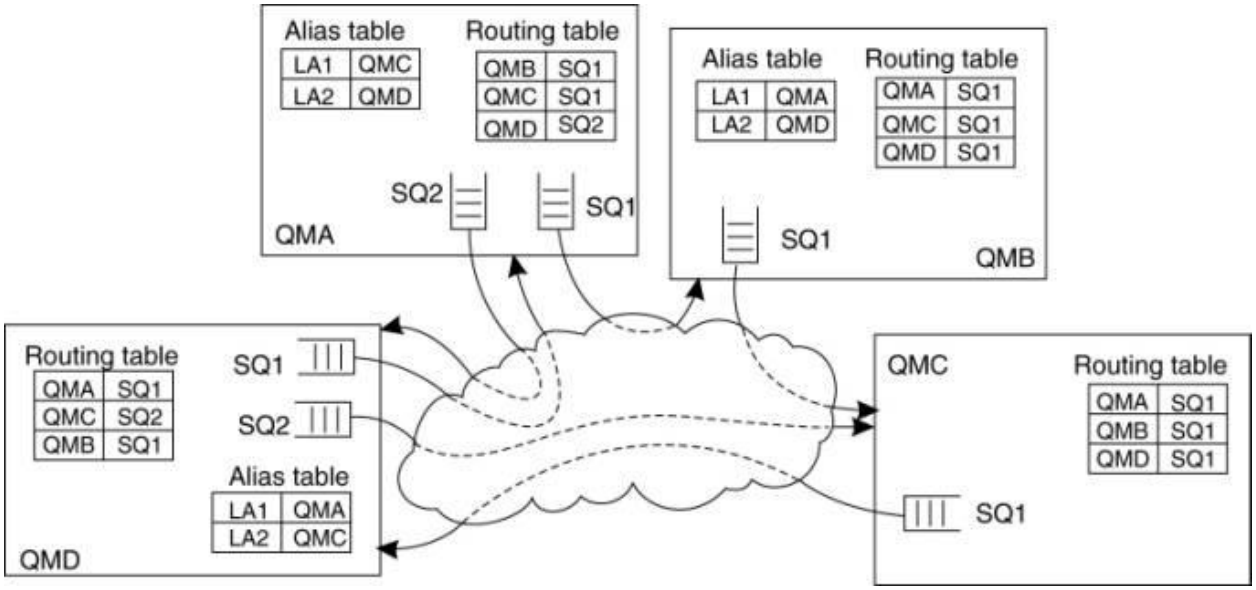
The principle of using routing tables and aliases is shown in Fig. 4-24. For example, an application linked to queue manager QMA can refer to a remote queue manager using the local alias LA1. The queue manager will first look up the actual destination in the alias table to find it is queue manager QMC. The route to QMC is found in the routing table, which states that messages for QMC should be appended to the outgoing queue SQ1, which is used to transfer messages to queue manager QMB. The latter will use its routing table to forward the message to QMC.

Figure 4-24. The general organization of an MQ queuing network using routing tables and aliases.
(This item is displayed on page 155 in the print version)



Following this approach of routing and aliasing leads to a programming interface that, fundamentally, is relatively simple, called the Message Queue Interface (MQI). The most important primitives of MQI are summarized in Fig. 4-25.

Figure 4-25. Primitives available in the message-queuing interface.

Primitive	Description
MQopen	Open a (possibly remote) queue
MQclose	Close a queue
MQput	Put a message into an opened queue
MQget	Get a message from a (local) queue

To put messages into a queue, an application calls the MQopen primitive, specifying a destination queue in a specific queue manager. The queue manager can be named using the locally-available alias. Whether the destination queue is actually remote or not is completely transparent to the application. MQopen should also be called if the application wants to get messages from its local queue. Only local queues can be opened for reading incoming messages. When an application is finished with accessing a queue, it should close it by calling MQclose.

Messages can be written to, or read from, a queue using MQput and MQget, respectively. In principle, messages are removed from a queue on a priority basis. Messages with the same priority are removed on a first-in, first-out basis, that is, the longest pending message is removed first. It is also possible to request for specific messages. Finally, MQ provides facilities to signal applications when messages have arrived, thus avoiding that an application will continuously have to poll a message queue for incoming messages.

Managing Overlay Networks

From the description so far, it should be clear that an important part of managing MQ systems is connecting the various queue managers into a consistent overlay network. Moreover, this network needs to be maintained over time. For small networks, this maintenance will not require much more than average administrative work, but matters become complicated when message queuing is used to integrate and disintegrate large existing systems.
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A major issue with MQ is that overlay networks need to be manually administrated. This administration not only involves creating channels between queue managers, but also filling in the routing tables. Obviously, this can grow into a nightmare. Unfortunately, management support for MQ systems is advanced only in the sense that an administrator can set virtually every possible attribute, and tweak any thinkable configuration. However, the bottom line is that channels and routing tables need to be manually maintained.

At the heart of overlay management is the channel control function component, which logically sits between message channel agents. This component allows an operator to monitor exactly what is going on at two end points of a channel. In addition, it is used to create channels and routing tables, but also to manage the queue managers that host the message channel agents. In a way, this approach to overlay management strongly resembles the management of cluster servers where a single administration server is used. In the latter case, the server essentially offers only a remote shell to each machine in the cluster, along with a few collective operations to handle groups of machines. The good news about distributed-systems management is that it offers lots of opportunities if you are looking for an area to explore new solutions to serious problems.

4.4. Stream-Oriented Communication

Communication as discussed so far has concentrated on exchanging more-or-less independent and complete units of information. Examples include a request for invoking a procedure, the reply to such a request, and messages exchanged between applications as in

message-queuing systems. The characteristic feature of this type of communication is that it does not matter at what particular point in time communication takes place. Although a system may perform too slow or too fast, timing has no effect on correctness.

There are also forms of communication in which timing plays a crucial role. Consider, for example, an audio stream built up as a sequence of 16-bit samples, each representing the amplitude of the sound wave as is done through Pulse Code Modulation (PCM). Also assume that the audio stream represents CD quality, meaning that the original sound wave has been sampled at a frequency of 44,100 Hz. To reproduce the original sound, it is essential that the samples in the audio stream are played out in the order they appear in the stream, but also at intervals of exactly 1/44,100 sec. Playing out at a different rate will produce an incorrect version of the original sound.

The question that we address in this section is which facilities a distributed system should offer to exchange time-dependent information such as audio and video streams. Various network protocols that deal with stream-oriented communication are discussed in Halsall (2001). Steinmetz and Nahrstedt (2004) provide an overall introduction to multimedia issues, part of which forms stream-oriented communication. Query processing on data streams is discussed in Babcock et al. (2002).
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4.4.1. Support for Continuous Media

Support for the exchange of time-dependent information is often formulated as support for continuous media. A medium refers to the means by which information is conveyed. These means include storage and transmission media, presentation media such as a monitor, and so on. An important type of medium is the way that information is represented. In other words, how is information encoded in a computer system? Different representations are used for different types of information. For example, text is generally encoded as ASCII or Unicode. Images can be represented in different formats such as GIF or JPEG. Audio streams can be encoded in a computer system by, for example, taking 16-bit samples using PCM.

In continuous (representation) media, the temporal relationships between different data items are fundamental to correctly interpreting what the data actually means. We already gave an example of reproducing a sound wave by playing out an audio stream. As another example, consider motion. Motion can be represented by a series of images in which successive images must be displayed at a uniform spacing T in time, typically 30–40 msec per image. Correct reproduction requires not only showing the stills in the correct order, but also at a constant frequency of 1/T images per second.

In contrast to continuous media, discrete (representation) media, is characterized by the fact that temporal relationships between data items are not fundamental to correctly interpreting the

data. Typical examples of discrete media include representations of text and still images, but also object code or executable files.

Data Stream

To capture the exchange of time-dependent information, distributed systems generally provide support for data streams. A data stream is nothing but a sequence of data units. Data streams can be applied to discrete as well as continuous media. For example, UNIX pipes or TCP/IP connections are typical examples of (byte-oriented) discrete data streams. Playing an audio file typically requires setting up a continuous data stream between the file and the audio device.

Timing is crucial to continuous data streams. To capture timing aspects, a distinction is often made between different transmission modes. In asynchronous transmission mode the data items in a stream are transmitted one after the other, but there are no further timing constraints on when transmission of items should take place. This is typically the case for discrete data streams. For example, a file can be transferred as a data stream, but it is mostly irrelevant exactly when the transfer of each item completes.
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In synchronous transmission mode, there is a maximum end-to-end delay defined for each unit in a data stream. Whether a data unit is transferred much faster than the maximum tolerated delay is not important. For example, a sensor may sample temperature at a certain rate and pass it through a network to an operator. In that case, it may be important that the end-to-end propagation time through the network is guaranteed to be lower than the time interval between taking samples, but it cannot do any harm if samples are propagated much faster than necessary.

Finally, in isochronous transmission mode, it is necessary that data units are transferred on time. This means that data transfer is subject to a maximum and minimum end-to-end delay, also referred to as bounded (delay) jitter. Isochronous transmission mode is particularly interesting for distributed multimedia systems, as it plays a crucial role in representing audio and video. In this chapter, we consider only continuous data streams using isochronous transmission, which we will refer to simply as streams.

Streams can be simple or complex. A simple stream consists of only a single sequence of data, whereas a complex stream consists of several related simple streams, called substreams. The relation between the substreams in a complex stream is often also time dependent. For example, stereo audio can be transmitted by means of a complex stream consisting of two substreams, each used for a single audio channel. It is important, however, that those two substreams are continuously synchronized. In other words, data units from each stream are to be communicated pairwise to ensure the effect of stereo. Another example of a complex stream is one for transmitting a movie. Such a stream could consist of a single video stream, along with two streams for transmitting the sound of the movie in stereo. A fourth stream might contain

subtitles for the deaf, or a translation into a different language than the audio. Again, synchronization of the substreams is important. If synchronization fails, reproduction of the movie fails. We return to stream synchronization below.

From a distributed systems perspective, we can distinguish several elements that are needed for supporting streams. For simplicity, we concentrate on streaming stored data, as opposed to streaming live data. In the latter case, data is captured in real time and sent over the network to recipients. The main difference between the two is that streaming live data leaves less opportunities for tuning a stream. Following Wu et al. (2001), we can then sketch a general client-server architecture for supporting continuous multimedia streams as shown in Fig. 4-26.

Figure 4-26. A general architecture for streaming stored multimedia data over a network.
(This item is displayed on page 160 in the print version)

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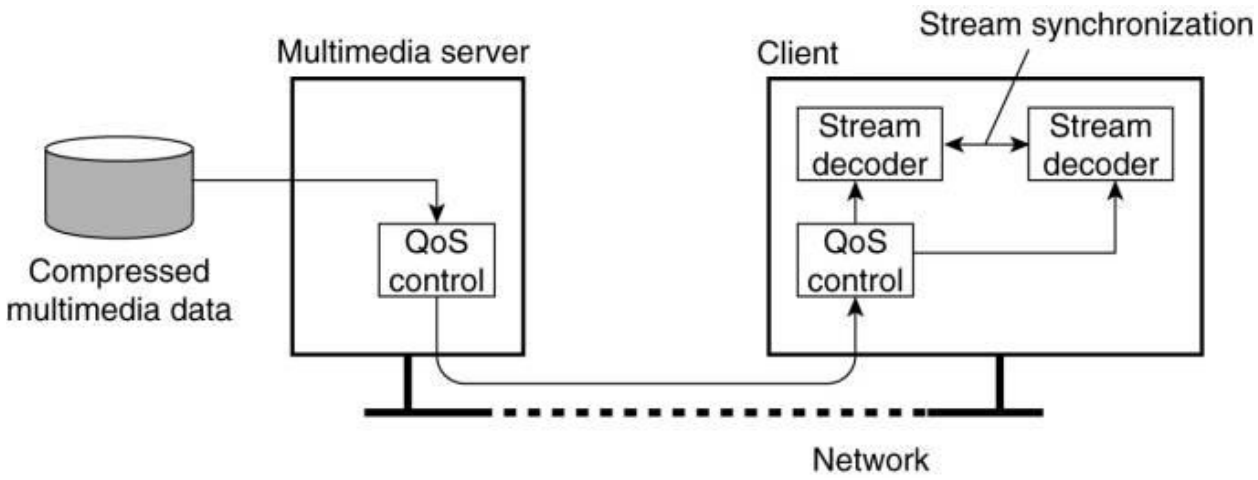

This general architecture reveals a number of important issues that need to be dealt with. In the first place, the multimedia data, notably video and to a lesser extent audio, will need to be compressed substantially in order to reduce the required storage and especially the network capacity. More important from the perspective of communication are controlling the quality of the transmission and synchronization issues. We discuss these issues next.
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4.4.2. Streams and Quality of Service

Timing (and other nonfunctional) requirements are generally expressed as Quality of Service (QoS) requirements. These requirements describe what is needed from the underlying

distributed system and network to ensure that, for example, the temporal relationships in a stream can be preserved. QoS for continuous data streams mainly concerns timeliness, volume, and reliability. In this section we take a closer look at QoS and its relation to setting up a stream.

Much has been said about how to specify required QoS (see, e.g., Jin and Nahrstedt, 2004). From an application's perspective, in many cases it boils down to specifying a few important properties (Halsall, 2001):

1. The required bit rate at which data should be transported.

2. The maximum delay until a session has been set up (i.e., when an application can start sending data).

3. The maximum end-to-end delay (i.e., how long it will take until a data unit makes it to a recipient).

4. The maximum delay variance, or jitter.

5. The maximum round-trip delay.

It should be noted that many refinements can be made to these specifications, as explained, for example, by Steinmetz and Nahrstadt (2004). However, when dealing with stream-oriented communication that is based on the Internet protocol stack, we simply have to live with the fact that the basis of communication is formed by an extremely simple, best-effort datagram service: IP. When the going gets tough, as may easily be the case in the Internet, the specification of IP allows a protocol implementation to drop packets whenever it sees fit. Many, if not all distributed systems that support stream-oriented communication, are currently built on top of the Internet protocol stack. So much for QoS specifications. (Actually, IP does provide some QoS support, but it is rarely implemented.)
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Enforcing QoS

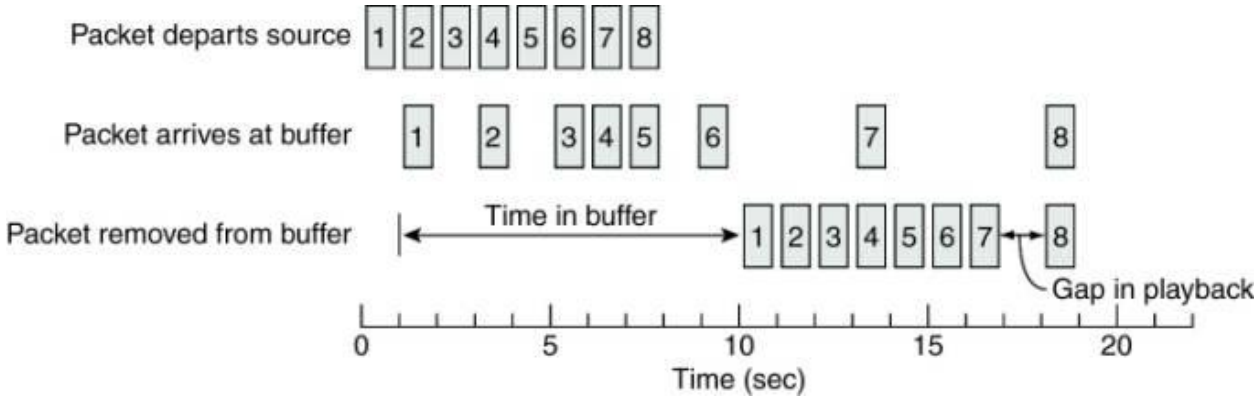
Given that the underlying system offers only a best-effort delivery service, a distributed system can try to conceal as much as possible of the lack of quality of service. Fortunately, there are several mechanisms that it can deploy.

First, the situation is not really so bad as sketched so far. For example, the Internet provides a means for differentiating classes of data by means of its differentiated services. A sending host can essentially mark outgoing packets as belonging to one of several classes, including an expedited forwarding class that essentially specifies that a packet should be forwarded by the current router with absolute priority (Davie et al., 2002). In addition, there is also an assured

forwarding class, by which traffic is divided into four subclasses, along with three ways to drop packets if the network gets congested. Assured forwarding therefore effectively defines a range of priorities that can be assigned to packets, and as such allows applications to differentiate time-sensitive packets from noncritical ones.

Besides these network-level solutions, a distributed system can also help in getting data across to receivers. Although there are generally not many tools available, one that is particularly useful is to use buffers to reduce jitter. The principle is simple, as shown in Fig. 4-27. Assuming that packets are delayed with a certain variance when transmitted over the network, the receiver simply stores them in a buffer for a maximum amount of time. This will allow the receiver to pass packets to the application at a regular rate, knowing that there will always be enough packets entering the buffer to be played back at that rate.

Figure 4-27. Using a buffer to reduce jitter.



Of course, things may go wrong, as is illustrated by packet #8 in Fig. 4-27. The size of the receiver's buffer corresponds to 9 seconds of packets to pass to the application. Unfortunately, packet #8 took 11 seconds to reach the receiver, at which time the buffer will have been completely emptied. The result is a gap in the playback at the application. The only solution is to increase the buffer size. The obvious drawback is that the delay at which the receiving application can start playing back the data contained in the packets increases as well.
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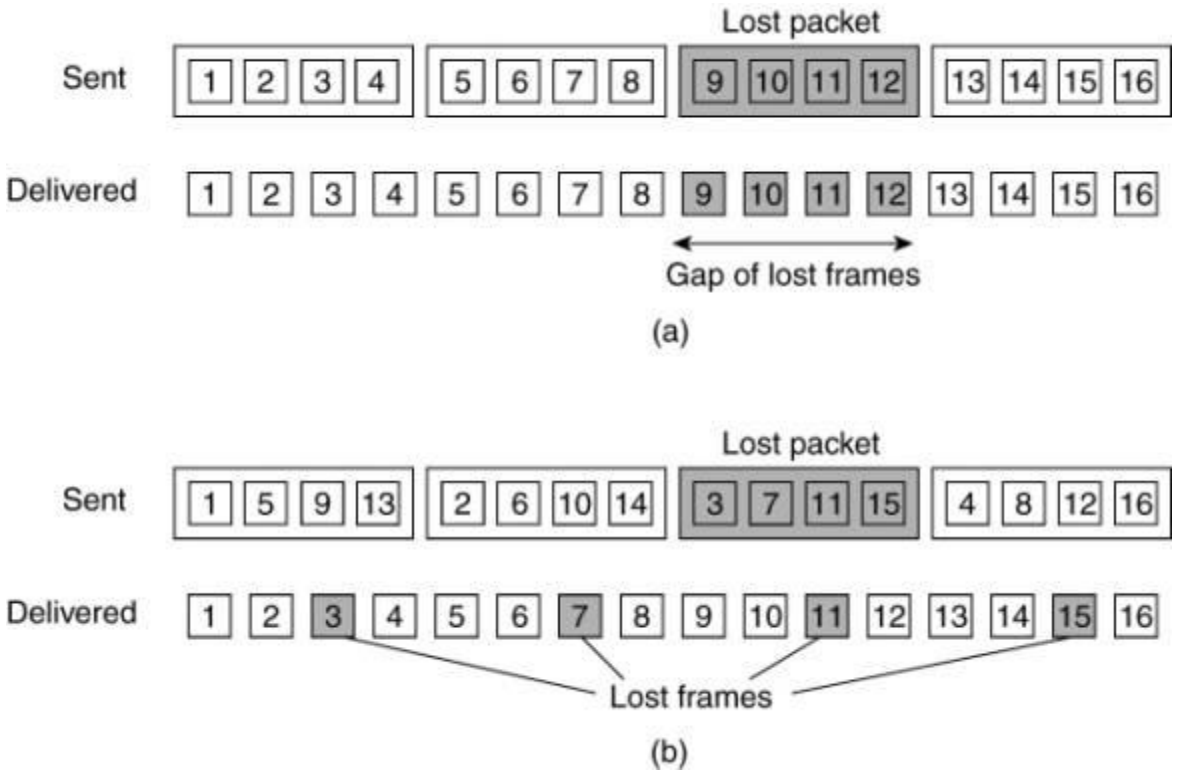
Other techniques can be used as well. Realizing that we are dealing with an underlying best-effort service also means that packets may be lost. To compensate for this loss in quality of service, we need to apply error correction techniques (Perkins et al., 1998; and Wah et al., 2000). Requesting the sender to retransmit a missing packet is generally out of the question, so that forward error correction (FEC) needs to be applied. A well-known technique is to encode the outgoing packets in such a way that any k out of n received packets is enough to reconstruct k correct packets.

One problem that may occur is that a single packet contains multiple audio and video frames. As a consequence, when a packet is lost, the receiver may actually perceive a large gap when playing out frames. This effect can be somewhat circumvented by interleaving frames, as shown in Fig. 4-28. In this way, when a packet is lost, the resulting gap in successive frames is distributed over time. Note, however, that this approach does require a larger receive buffer in comparison to noninterleaving, and thus imposes a higher start delay for the receiving application. For example, when considering Fig. 4-28(b), to play the first four frames, the receiver will need to have four packets delivered, instead of only one packet in comparison to noninterleaved transmission.

Figure 4-28. The effect of packet loss in (a) noninterleaved transmission and (b) interleaved transmission.



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4.4.3. Stream Synchronization

An important issue in multimedia systems is that different streams, possibly in the form of a complex stream, are mutually synchronized. Synchronization of streams deals with maintaining temporal relations between streams. Two types of synchronization occur.

The simplest form of synchronization is that between a discrete data stream and a continuous data stream. Consider, for example, a slide show on the Web that has been enhanced with audio. Each slide is transferred from the server to the client in the form of a discrete data stream. At the same time, the client should play out a specific (part of an) audio stream that matches the current slide that is also fetched from the server. In this case, the audio stream is to be synchronized with the presentation of slides.

A more demanding type of synchronization is that between continuous data streams. A daily example is playing a movie in which the video stream needs to be synchronized with the audio, commonly referred to as lip synchronization. Another example of synchronization is playing a stereo audio stream consisting of two substreams, one for each channel. Proper play out requires that the two substreams are tightly synchronized: a difference of more than 20 μsec can distort the stereo effect.

Synchronization takes place at the level of the data units of which a stream is made up. In other words, we can synchronize two streams only between data units. The choice of what exactly a data unit is depends very much on the level of abstraction at which a data stream is viewed. To make things concrete, consider again a CD-quality (single-channel) audio stream. At the finest granularity, such a stream appears as a sequence of 16-bit samples. With a sampling frequency of 44,100 Hz, synchronization with other audio streams could, in theory, take place approximately every 23 μsec. For high-quality stereo effects, it turns out that synchronization at this level is indeed necessary.

However, when we consider synchronization between an audio stream and a video stream for lip synchronization, a much coarser granularity can be taken. As we explained, video frames need to be displayed at a rate of 25 Hz or more. Taking the widely-used NTSC standard of 29.97 Hz, we could group audio samples into logical units that last as long as a video frame is displayed (33 msec). With an audio sampling frequency of 44,100 Hz, an audio data unit can thus be as large as 1470 samples, or 11,760 bytes (assuming each sample is 16 bits). In practice, larger units lasting 40 or even 80 msec can be tolerated (Steinmetz, 1996).

Synchronization Mechanisms

Let us now see how synchronization is actually done. Two issues need to be distinguished: (1) the basic mechanisms for synchronizing two streams, and (2) the distribution of those mechanisms in a networked environment.

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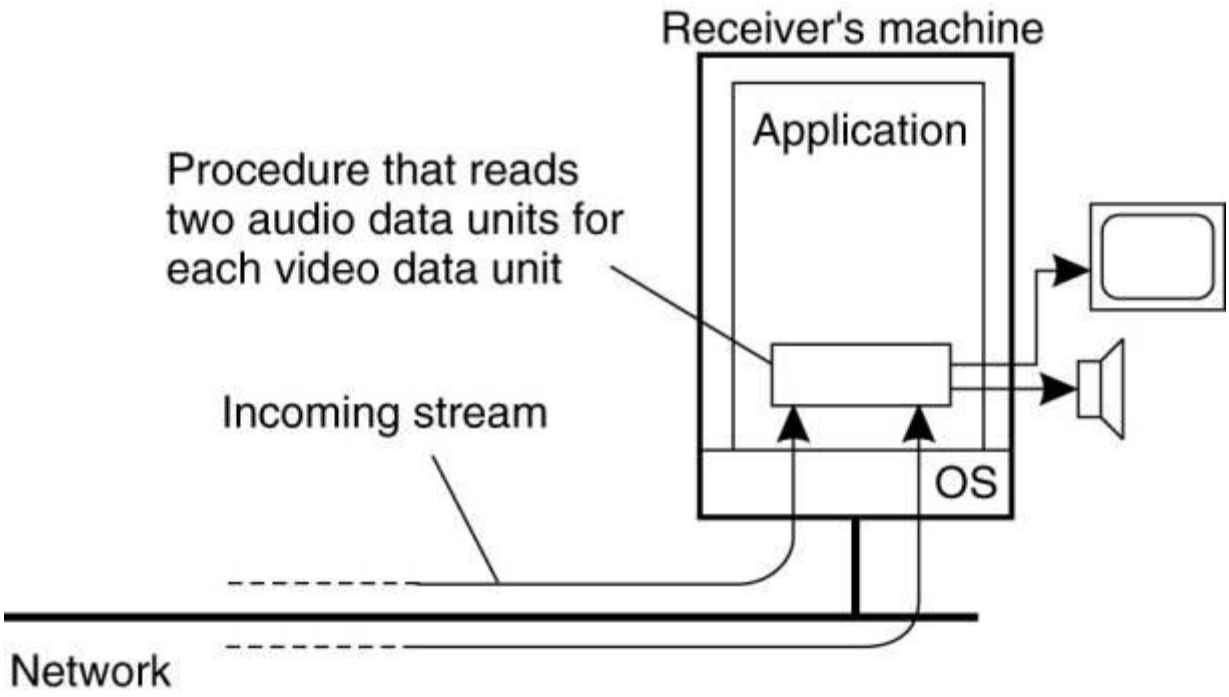
Synchronization mechanisms can be viewed at several different levels of abstraction. At the lowest level, synchronization is done explicitly by operating on the data units of simple streams. This principle is shown in Fig. 4-29. In essence, there is a process that simply executes read and write operations on several simple streams, ensuring that those operations adhere to specific timing and synchronization constraints.

Figure 4-29. The principle of explicit synchronization on the level data units.



For example, consider a movie that is presented as two input streams. The video stream contains uncompressed low-quality images of 320x240 pixels, each encoded by a single byte, leading to video data units of 76,800 bytes each. Assume that images are to be displayed at 30 Hz, or one image every 33 msec. The audio stream is assumed to contain audio samples grouped into units of 11760 bytes, each corresponding to 33 ms of audio, as explained above. If the input process can handle 2.5 MB/sec, we can achieve lip synchronization by simply alternating between reading an image and reading a block of audio samples every 33 ms.

The drawback of this approach is that the application is made completely responsible for implementing synchronization while it has only low-level facilities available. A better approach is to offer an application an interface that allows it to more easily control streams and devices. Returning to our example, assume that the video display has a control interface that allows it to specify the rate at which images should be displayed. In addition, the interface offers the facility to register a user-defined handler that is called each time k new images have arrived. An analogous interface is offered by the audio device. With these control interfaces, an application developer can write a simple monitor program consisting of two handlers, one for each stream, that jointly check if the video and audio stream are sufficiently synchronized, and if necessary, adjust the rate at which video or audio units are presented.

This last example is illustrated in Fig. 4-30, and is typical for many multimedia middleware systems. In effect, multimedia middleware offers a collection of interfaces for controlling audio and video streams, including interfaces for controlling devices such as monitors, cameras,

microphones, etc. Each device and stream has its own high-level interfaces, including interfaces for notifying an application when some event occurred. The latter are subsequently used to write handlers for synchronizing streams. Examples of such interfaces are given in Blair and Stefani (1998).
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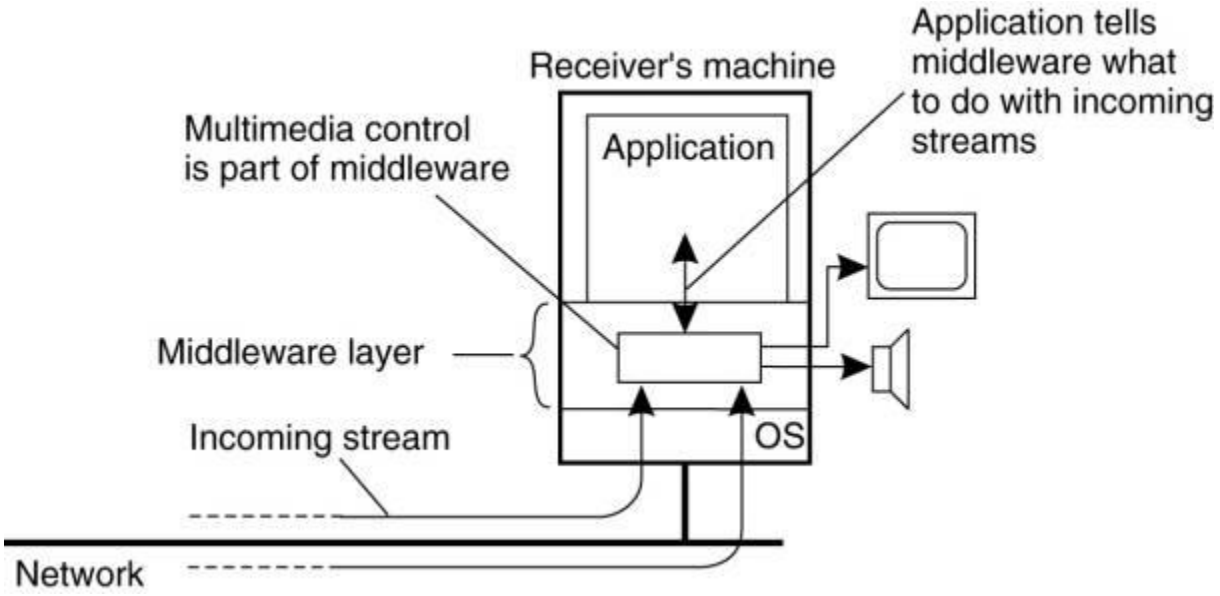
Figure 4-30. The principle of synchronization as supported by high-level interfaces.



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The distribution of synchronization mechanisms is another issue that needs to be looked at. First, the receiving side of a complex stream consisting of substreams that require synchronization, needs to know exactly what to do. In other words, it must have a complete synchronization specification locally available. Common practice is to provide this information implicitly by multiplexing the different streams into a single stream containing all data units, including those for synchronization.

This latter approach to synchronization is followed for MPEG streams. The MPEG (Motion Picture Experts Group) standards form a collection of algorithms for compressing video and audio. Several MPEG standards exist. MPEG-2, for example, was originally designed for compressing broadcast quality video into 4 to 6 Mbps. In MPEG-2, an unlimited number of continuous and discrete streams can be merged into a single stream. Each input stream is first turned into a stream of packets that carry a timestamp based on a 90-kHz system clock. These streams are subsequently multiplexed into a program stream then consisting of variable-length packets, but which have in common that they all have the same time base. The receiving side demultiplexes the stream, again using the timestamps of each packet as the basic mechanism for interstream synchronization.

Another important issue is whether synchronization should take place at the sending or the receiving side. If the sender handles synchronization, it may be possible to merge streams into a single stream with a different type of data unit. Consider again a stereo audio stream consisting of two substreams, one for each channel. One possibility is to transfer each stream independently to the receiver and let the latter synchronize the samples pairwise. Obviously, as each substream may be subject to different delays, synchronization can be extremely difficult. A better approach is to merge the two substreams at the sender. The resulting stream consists of data units consisting of pairs of samples, one for each channel. The receiver now merely has to read in a data unit, and split it into a left and right sample. Delays for both channels are now identical.

4.5. Multicast Communication

An important topic in communication in distributed systems is the support for sending data to multiple receivers, also known as multicast communication. For many years, this topic has belonged to the domain of network protocols, where numerous proposals for network-level and transport-level solutions have been implemented and evaluated (Janic, 2005; and Obraczka, 1998). A major issue in all solutions was setting up the communication paths for information dissemination. In practice, this involved a huge management effort, in many cases requiring human intervention. In addition, as long as there is no convergence of proposals, ISPs have shown to be reluctant to support multicasting (Diot et al., 2000).

With the advent of peer-to-peer technology, and notably structured overlay management, it became easier to set up communication paths. As peer-to-peer solutions are typically deployed at the application layer, various application-level multicasting techniques have been introduced. In this section, we will take a brief look at these techniques.

Multicast communication can also be accomplished in other ways than setting up explicit communication paths. As we also explore in this section, gossip-based information dissemination provides simple (yet often less efficient) ways for multicasting.

4.5.1. Application-Level Multicasting

The basic idea in application-level multicasting is that nodes organize into an overlay network, which is then used to disseminate information to its members. An important observation is that network routers are not involved in group membership. As a consequence, the connections between nodes in the overlay network may cross several physical links, and as such, routing messages within the overlay may not be optimal in comparison to what could have been achieved by network-level routing.

A crucial design issue is the construction of the overlay network. In essence, there are two approaches (El-Sayed, 2003). First, nodes may organize themselves directly into a tree,

meaning that there is a unique (overlay) path between every pair of nodes. An alternative approach is that nodes organize into a mesh network in which every node will have multiple neighbors and, in general, there exist multiple paths between every pair of nodes. The main difference between the two is that the latter generally provides higher robustness: if a connection breaks (e.g., because a node fails), there will still be an opportunity to disseminate information without having to immediately reorganize the entire overlay network.
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To make matters concrete, let us consider a relatively simple scheme for constructing a multicast tree in Chord, which we described in Chap. 2. This scheme was originally proposed for Scribe (Castro et al., 2002) which is an application-level multicasting scheme built on top of Pastry (Rowstron and Druschel, 2001). The latter is also a DHT-based peer-to-peer system.

Assume a node wants to start a multicast session. To this end, it simply generates a multicast identifier, say mid which is just a randomly-chosen 160-bit key. It then looks up succ(mid), which is the node responsible for that key, and promotes it to become the root of the multicast tree that will be used to sending data to interested nodes. In order to join the tree, a node P simply executes the operation LOOKUP(mid) having the effect that a lookup message with the request to join the multicast group mid will be routed from P to succ(mid). As we mentioned before, the routing algorithm itself will be explained in detail in Chap. 5.

On its way toward the root, the join request will pass several nodes. Assume it first reaches node Q. If Q had never seen a join request for mid before, it will become a forwarder for that group. At that point, P will become a child of Q whereas the latter will continue to forward the join request to the root. If the next node on the root, say R is also not yet a forwarder, it will become one and record Q as its child and continue to send the join request.

On the other hand, if Q (or R) is already a forwarder for mid, it will also record the previous sender as its child (i.e., P or Q, respectively), but there will not be a need to send the join request to the root anymore, as Q (or R) will already be a member of the multicast tree.

Nodes such as P that have explicitly requested to join the multicast tree are, by definition, also forwarders. The result of this scheme is that we construct a multicast tree across the overlay network with two types of nodes: pure forwarders that act as helpers, and nodes that are also forwarders, but have explicitly requested to join the tree. Multicasting is now simple: a node merely sends a multicast message toward the root of the tree by again executing the LOOKUP(mid) operation, after which that message can be sent along the tree.

We note that this high-level description of multicasting in Scribe does not do justice to its original design. The interested reader is therefore encouraged to take a look at the details, which can be found in Castro et al. (2002).

Overlay Construction

From the high-level description given above, it should be clear that although building a tree by itself is not that difficult once we have organized the nodes into an overlay, building an efficient tree may be a different story. Note that in our description so far, the selection of nodes that participate in the tree does not take into account any performance metrics: it is purely based on the (logical) routing of messages through the overlay.
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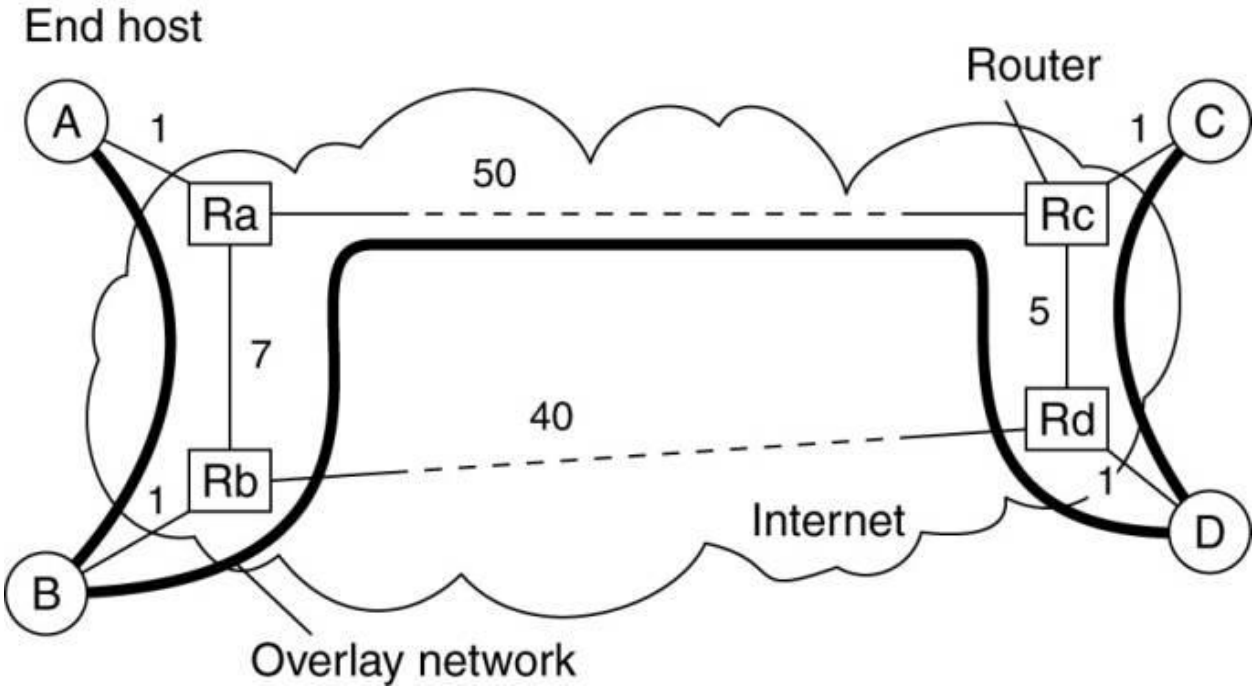
To understand the problem at hand, take a look at Fig. 4-31 which shows a small set of four nodes that are organized in a simple overlay network, with node A forming the root of a multicast tree. The costs for traversing a physical link are also shown. Now, whenever A multicasts a message to the other nodes, it is seen that this message will traverse each of the links <B, Rb>, <Ra, Rb>, <Rc, Rd>, and <D, Rd> twice. The overlay network would have been more efficient if we had not constructed an overlay link from B to D, but instead from A to C. Such a configuration would have saved the double traversal across links <Ra, Rb> and <Rc, Rd>.

Figure 4-31. The relation between links in an overlay and actual network-level routes.



The quality of an application-level multicast tree is generally measured by three different metrics: link stress, stretch, and tree cost. Link stress is defined per link and counts how often a packet crosses the same link (Chu et al., 2002). A link stress greater than 1 comes from the fact that although at a logical level a packet may be forwarded along two different connections, part

of those connections may actually correspond to the same physical link, as we showed in Fig. 4-31.

The stretch or Relative Delay Penalty (RDP) measures the ratio in the delay between two nodes in the overlay, and the delay that those two nodes would experience in the underlying network. For example, in the overlay network, messages from B to C follow the route B Rb Ra Rc C, having a total cost of 59 units. However, messages would have been routed in the underlying network along the path B Rb Rd Rc C, with a total cost of 47 units, leading to a stretch of 1.255. Obviously, when constructing an overlay network, the goal is to minimize the aggregated stretch, or similarly, the average RDP measured over all node pairs.

Finally, the tree cost is a global metric, generally related to minimizing the aggregated link costs. For example, if the cost of a link is taken to be the delay between its two end nodes, then optimizing the tree cost boils down to finding a minimal spanning tree in which the total time for disseminating information to all nodes is minimal.
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To simplify matters somewhat, assume that a multicast group has an associated and well-known node that keeps track of the nodes that have joined the tree. When a new node issues a join request, it contacts this rendezvous node to obtain a (potentially partial) list of members. The goal is to select the best member that can operate as the new node's parent in the tree. Who should it select? There are many alternatives and different proposals often follow very different solutions.

Consider, for example, a multicast group with only a single source. In this case, the selection of the best node is obvious: it should be the source (because in that case we can be assured that the stretch will be equal to 1). However, in doing so, we would introduce a star topology with the source in the middle. Although simple, it is not difficult to imagine the source may easily become overloaded. In other words, selection of a node will generally be constrained in such a way that only those nodes may be chosen who have k or less neighbors, with k being a design parameter. This constraint severely complicates the tree-establishment algorithm, as a good solution may require that part of the existing tree is reconfigured.

Tan et al. (2003) provide an extensive overview and evaluation of various solutions to this problem. As an illustration, let us take a closer look at one specific family, known as switch-trees (Helder and Jamin, 2002). The basic idea is simple. Assume we already have a multicast tree with a single source as root. In this tree, a node P can switch parents by dropping the link to its current parent in favor of a link to another node. The only constraints imposed on switching links is that the new parent can never be a member of the subtree rooted at P (as this would partition the tree and create a loop), and that the new parent will not have too many immediate children. The latter is needed to limit the load of forwarding messages by any single node.

There are different criteria for deciding to switch parents. A simple one is to optimize the route to the source, effectively minimizing the delay when a message is to be multicast. To this end,

each node regularly receives information on other nodes (we will explain one specific way of doing this below). At that point, the node can evaluate whether another node would be a better parent in terms of delay along the route to the source, and if so, initiates a switch.

Another criteria could be whether the delay to the potential other parent is lower than to the current parent. If every node takes this as a criterion, then the aggregated delays of the resulting tree should ideally be minimal. In other words, this is an example of optimizing the cost of the tree as we explained above. However, more information would be needed to construct such a tree, but as it turns out, this simple scheme is a reasonable heuristic leading to a good approximation of a minimal spanning tree.

As an example, consider the case where a node P receives information on the neighbors of its parent. Note that the neighbors consist of P's grandparent, along with the other siblings of P's parent. Node P can then evaluate the delays to each of these nodes and subsequently choose the one with the lowest delay, say Q, as its new parent. To that end, it sends a switch request to Q. To prevent loops from being formed due to concurrent switching requests, a node that has an outstanding switch request will simply refuse to process any incoming requests. In effect, this leads to a situation where only completely independent switches can be carried out simultaneously. Furthermore, P will provide Q with enough information to allow the latter to conclude that both nodes have the same parent, or that Q is the grandparent.
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An important problem that we have not yet addressed is node failure. In the case of switch-trees, a simple solution is proposed: whenever a node notices that its parent has failed, it simply attaches itself to the root. At that point, the optimization protocol can proceed as usual and will eventually place the node at a good point in the multicast tree. Experiments described in Helder and Jamin (2002) show that the resulting tree is indeed close to a minimal spanning one.

4.5.2. Gossip-Based Data Dissemination

An increasingly important technique for disseminating information is to rely on epidemic behavior. Observing how diseases spread among people, researchers have since long investigated whether simple techniques could be developed for spreading information in very large-scale distributed systems. The main goal of these epidemic protocols is to rapidly propagate information among a large collection of nodes using only local information. In other words, there is no central component by which information dissemination is coordinated.

To explain the general principles of these algorithms, we assume that all updates for a specific data item are initiated at a single node. In this way, we simply avoid write-write conflicts. The following presentation is based on the classical paper by Demers et al. (1987) on epidemic algorithms. A recent overview of epidemic information dissemination can be found in Eugster at el. (2004).

Information Dissemination Models

As the name suggests, epidemic algorithms are based on the theory of epidemics, which studies the spreading of infectious diseases. In the case of large-scale distributed systems, instead of spreading diseases, they spread information. Research on epidemics for distributed systems also aims at a completely different goal: whereas health organizations will do their utmost best to prevent infectious diseases from spreading across large groups of people, designers of epidemic algorithms for distributed systems will try to "infect" all nodes with new information as fast as possible.

Using the terminology from epidemics, a node that is part of a distributed system is called infected if it holds data that it is willing to spread to other nodes. A node that has not yet seen this data is called susceptible. Finally, an updated node that is not willing or able to spread its data is said to be removed. Note that we assume we can distinguish old from new data, for example, because it has been timestamped or versioned. In this light, nodes are also said to spread updates.
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A popular propagation model is that of anti-entropy. In this model, a node P picks another node Q at random, and subsequently exchanges updates with Q. There are three approaches to exchanging updates:

1. P only pushes its own updates to Q

2. P only pulls in new updates from Q

3. P and Q send updates to each other (i.e., a push-pull approach)

When it comes to rapidly spreading updates, only pushing updates turns out to be a bad choice. Intuitively, this can be understood as follows. First, note that in a pure push-based approach, updates can be propagated only by infected nodes. However, if many nodes are infected, the probability of each one selecting a susceptible node is relatively small. Consequently, chances are that a particular node remains susceptible for a long period simply because it is not selected by an infected node.

In contrast, the pull-based approach works much better when many nodes are infected. In that case, spreading updates is essentially triggered by susceptible nodes. Chances are large that such a node will contact an infected one to subsequently pull in the updates and become infected as well.

It can be shown that if only a single node is infected, updates will rapidly spread across all nodes using either form of anti-entropy, although push-pull remains the best strategy (Jelasity et al., 2005a). Define a round as spanning a period in which every node will at least once have taken the initiative to exchange updates with a randomly chosen other node. It can then be

shown that the number of rounds to propagate a single update to all nodes takes O(log (N)) rounds, where N is the number of nodes in the system. This indicates indeed that propagating updates is fast, but above all scalable.

One specific variant of this approach is called rumor spreading, or simply gossiping. It works as follows. If node P has just been updated for data item x, it contacts an arbitrary other node Q and tries to push the update to Q. However, it is possible that Q was already updated by another node. In that case, P may lose interest in spreading the update any further, say with probability 1/k. In other words, it then becomes removed.

Gossiping is completely analogous to real life. When Bob has some hot news to spread around, he may phone his friend Alice telling her all about it. Alice, like Bob, will be really excited to spread the gossip to her friends as well. However, she will become disappointed when phoning a friend, say Chuck, only to hear that the news has already reached him. Chances are that she will stop phoning other friends, for what good is it if they already know?
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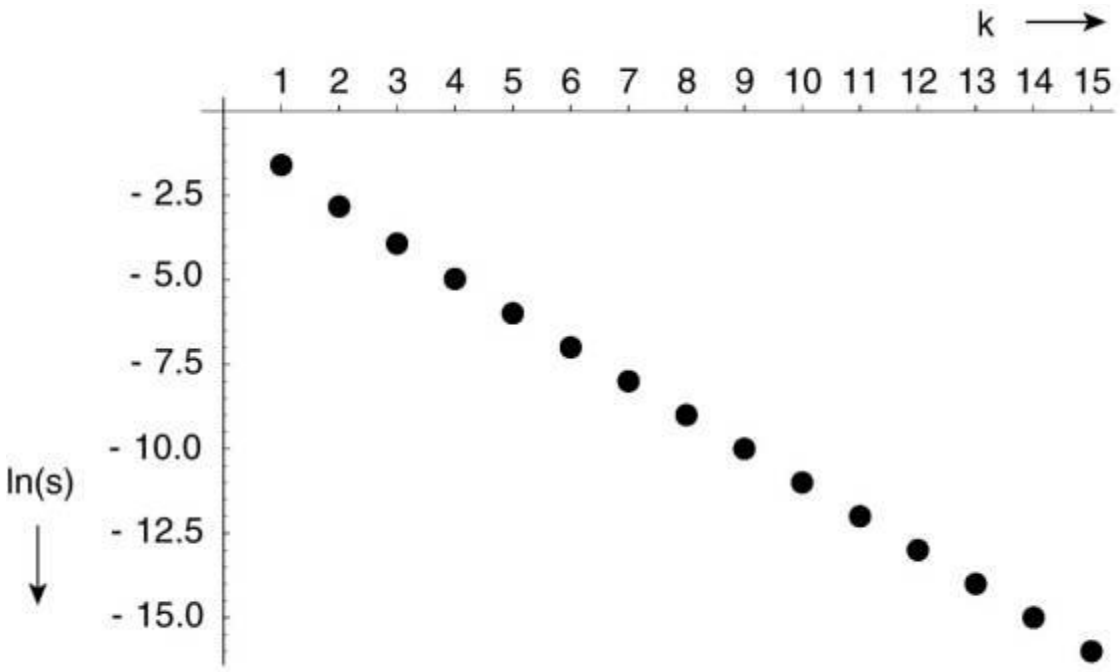
Gossiping turns out to be an excellent way of rapidly spreading news. However, it cannot guarantee that all nodes will actually be updated (Demers et al., 1987). It can be shown that when there is a large number of nodes that participate in the epidemics, the fraction s of nodes that will remain ignorant of an update, that is, remain susceptible, satisfies the equation:

$$s = e^{-(k + 1)(1 - s)}$$

Fig. 4-32 shows ln (s) as a function of k. For example, if k = 4, ln (s)=-4.97, so that s is less than 0.007, meaning that less than 0.7% of the nodes remain susceptible. Nevertheless, special measures are needed to guarantee that those nodes will also be updated. Combining anti-entropy with gossiping will do the trick.

Figure 4-32. The relation between the fraction s of update-ignorant nodes and the parameter k in pure gossiping. The graph displays ln(s) as a function of k.



One of the main advantages of epidemic algorithms is their scalability, due to the fact that the number of synchronizations between processes is relatively small compared to other propagation methods. For wide-area systems, Lin and Marzullo (1999) show that it makes sense to take the actual network topology into account to achieve better results. In their approach, nodes that are connected to only a few other nodes are contacted with a relatively high probability. The underlying assumption is that such nodes form a bridge to other remote parts of the network; therefore, they should be contacted as soon as possible. This approach is referred to as directional gossiping and comes in different variants.

This problem touches upon an important assumption that most epidemic solutions make, namely that a node can randomly select any other node to gossip with. This implies that, in principle, the complete set of nodes should be known to each member. In a large system, this assumption can never hold.
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Fortunately, there is no need to have such a list. As we explained in Chap. 2, maintaining a partial view that is more or less continuously updated will organize the collection of nodes into a random graph. By regularly updating the partial view of each node, random selection is no longer a problem.

Removing Data

Epidemic algorithms are extremely good for spreading updates. However, they have a rather strange side-effect: spreading the deletion of a data item is hard. The essence of the problem lies in the fact that deletion of a data item destroys all information on that item. Consequently, when a data item is simply removed from a node, that node will eventually receive old copies of the data item and interpret those as updates on something it did not have before.

The trick is to record the deletion of a data item as just another update, and keep a record of that deletion. In this way, old copies will not be interpreted as something new, but merely treated as versions that have been updated by a delete operation. The recording of a deletion is done by spreading death certificates.

Of course, the problem with death certificates is that they should eventually be cleaned up, or otherwise each node will gradually build a huge local database of historical information on deleted data items that is otherwise not used. Demers et al. (1987) propose to use what they call dormant death certificates. Each death certificate is timestamped when it is created. If it can be assumed that updates propagate to all nodes within a known finite time, then death certificates can be removed after this maximum propagation time has elapsed.

However, to provide hard guarantees that deletions are indeed spread to all nodes, only a very few nodes maintain dormant death certificates that are never thrown away. Assume node P has such a certificate for data item x. If by any chance an obsolete update for x reaches P, P will react by simply spreading the death certificate for x again.
Applications

To finalize this presentation, let us take a look at some interesting applications of epidemic protocols. We already mentioned spreading updates, which is perhaps the most widely-deployed application. Also, in Chap. 2 we discussed how providing positioning information about nodes can assist in constructing specific topologies. In the same light, gossiping can be used to discover nodes that have a few outgoing wide-area links, to subsequently apply directional gossiping as we mentioned above.

Another interesting application area is simply collecting, or actually aggregating information (Jelasity et al., 2005b). Consider the following information exchange. Every node i initially chooses an arbitrary number, say xi. When node i contacts node j, they each update their value as:
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xi,xj (xi + xj) /2

Obviously, after this exchange, both i and j will have the same value. In fact, it is not difficult to see that eventually all nodes will have the same value, namely the average of all initial values. Propagation speed is again exponential.

What use does computing the average have? Consider the situation that all nodes i have set xi to zero, except for x1, which has set it to 1:

If there N nodes, then eventually each node will compute the average, which is 1/N. As a consequence, every node i can estimate the size of the system as being 1/xi. This information alone can be used to dynamically adjust various system parameters. For example, the size of the partial view (i.e., the number of neighbors that each nodes keeps track of) should be dependent on the total number of participating nodes. Knowing this number will allow a node to dynamically adjust the size of its partial view. Indeed, this can be viewed as a property of self-management.

Computing the average may prove to be difficult when nodes regularly join and leave the system. One practical solution to this problem is to introduce epochs. Assuming that node 1 is stable, it simply starts a new epoch now and then. When node i sees a new epoch for the first time, it resets its own variable xi to zero and starts computing the average again.

Of course, other results can also be computed. For example, instead of having a fixed node (x1) start the computation of the average, we can easily pick a random node as follows. Every node i initially sets xi to a random number from the same interval, say [0,1], and also stores it permanently as mi. Upon an exchange between nodes i and j, each change their value to:

xi,xj max (xi,xj)

Each node i for which mi < xi will lose the competition for being the initiator in starting the computation of the average. In the end, there will be a single winner. Of course, although it is easy to conclude that a node has lost, it is much more difficult to decide that it has won, as it remains uncertain whether all results have come in. The solution to this problem is to be optimistic: a node always assumes it is the winner until proven otherwise. At that point, it simply resets the variable it is using for computing the average to zero. Note that by now, several different computations (in our example computing a maximum and computing an average) may be executing concurrently.