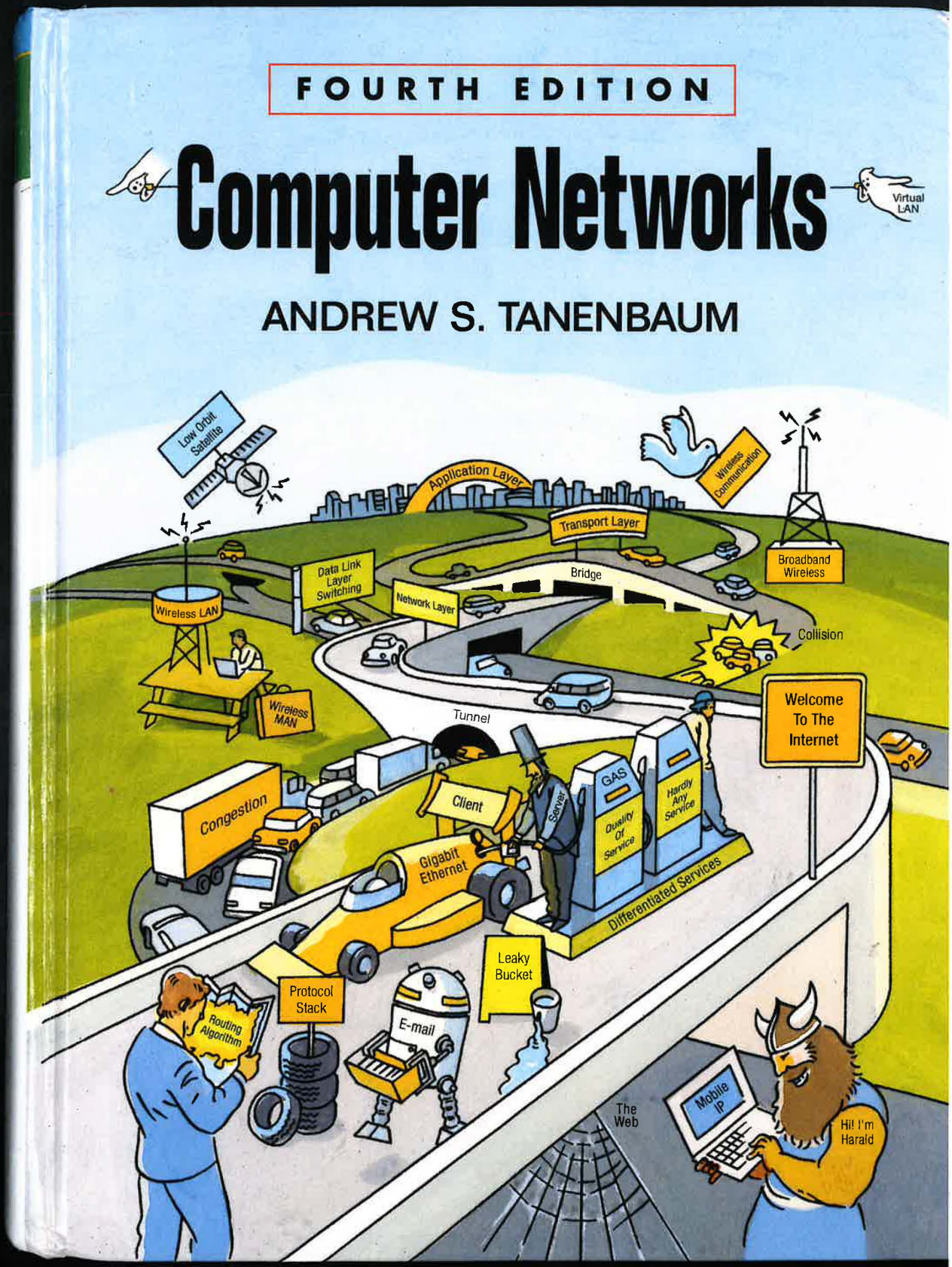
FOURTH EDITION

Computer Networks

ANDREW S. TANENBAUM



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Amsterdam, The Netherlands

Prentice Hall PTR
Upper Saddle River, NJ 07458
www.phptr.com

Library of Congress Cataloging-in-Publication Data

Tanenbaum, Andrew S.
Computer networks / Andrew S. Tanenbaum.--4th ed.
p. cm.
Includes bibliographical references.
ISBN 0-13-066102-3
1. Computer networks. I. Title.
TK5105.5 .T36 2002
004.6--dc21 2002029263

Editorial/production supervision: *Patti Guerrieri*
Cover design director: *Jerry Votta*
Cover designer: *Anthony Gemmellaro*
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Interior graphics: *Hadel Studio*
Typesetting: *Andrew S. Tanenbaum*
Manufacturing buyer: *Maura Zaldivar*
Executive editor: *Mary Franz*
Editorial assistant: *Noreen Regina*
Marketing manager: *Dan DePasquale*

Publishing as Prentice Hall PTR
Upper Saddle River, New Jersey 07458

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Printed in the United States of America

10 9 8 7 6 5 4 3

ISBN 0-13-066102-3

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Pearson Education Australia PTY, Limited
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connection record, are listed. If an action involves waking up a sleeping process, the actions following the wakeup also count. For example, if a CALL REQUEST packet comes in and a process was asleep waiting for it, the transmission of the CALL ACCEPT packet following the wakeup counts as part of the action for CALL REQUEST. After each action is performed, the connection may move to a new state, as shown in Fig. 6-21.

The advantage of representing the protocol as a matrix is threefold. First, in this form it is much easier for the programmer to systematically check each combination of state and event to see if an action is required. In production implementations, some of the combinations would be used for error handling. In Fig. 6-21 no distinction is made between impossible situations and illegal ones. For example, if a connection is in *waiting* state, the DISCONNECT event is impossible because the user is blocked and cannot execute any primitives at all. On the other hand, in *sending* state, data packets are not expected because no credit has been issued. The arrival of a data packet is a protocol error.

The second advantage of the matrix representation of the protocol is in implementing it. One could envision a two-dimensional array in which element $a[i][j]$ was a pointer or index to the procedure that handled the occurrence of event i when in state j. One possible implementation is to write the transport entity as a short loop, waiting for an event at the top of the loop. When an event happens, the relevant connection is located and its state is extracted. With the event and state now known, the transport entity just indexes into the array a and calls the proper procedure. This approach gives a much more regular and systematic design than our transport entity.

The third advantage of the finite state machine approach is for protocol description. In some standards documents, the protocols are given as finite state machines of the type of Fig. 6-21. Going from this kind of description to a working transport entity is much easier if the transport entity is also driven by a finite state machine based on the one in the standard.

The primary disadvantage of the finite state machine approach is that it may be more difficult to understand than the straight programming example we used initially. However, this problem may be partially solved by drawing the finite state machine as a graph, as is done in Fig. 6-22.

6.4 THE INTERNET TRANSPORT PROTOCOLS: UDP

The Internet has two main protocols in the transport layer, a connectionless protocol and a connection-oriented one. In the following sections we will study both of them. The connectionless protocol is UDP. The connection-oriented protocol is TCP. Because UDP is basically just IP with a short header added, we will start with it. We will also look at two applications of UDP.

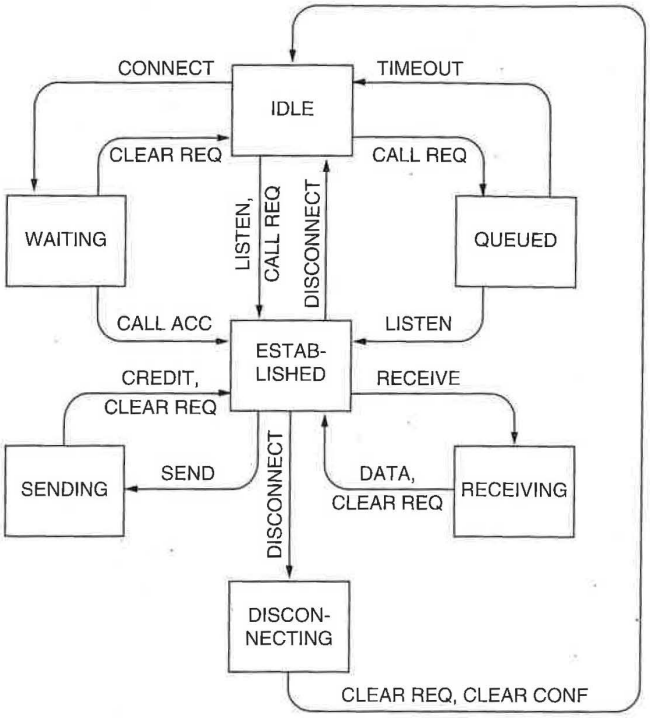

Figure 6-22. The example protocol in graphical form. Transitions that leave the connection state unchanged have been omitted for simplicity.

6.4.1 Introduction to UDP

The Internet protocol suite supports a connectionless transport protocol, **UDP** (**User Datagram Protocol**). UDP provides a way for applications to send encapsulated IP datagrams and send them without having to establish a connection. UDP is described in RFC 768.

UDP transmits **segments** consisting of an 8-byte header followed by the payload. The header is shown in Fig. 6-23. The two ports serve to identify the end points within the source and destination machines. When a UDP packet arrives, its payload is handed to the process attached to the destination port. This attachment occurs when BIND primitive or something similar is used, as we saw in Fig. 6-6 for TCP (the binding process is the same for UDP). In fact, the main value of having UDP over just using raw IP is the addition of the source and destination ports. Without the port fields, the transport layer would not know what to do with the packet. With them, it delivers segments correctly.

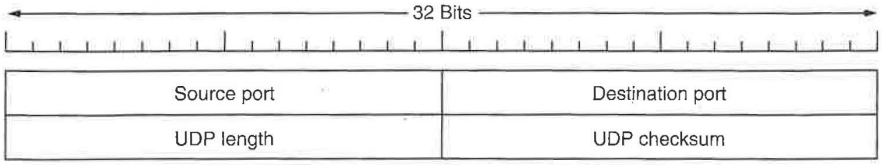

Figure 6-23. The UDP header.

The source port is primarily needed when a reply must be sent back to the source. By copying the *source port* field from the incoming segment into the *destination port* field of the outgoing segment, the process sending the reply can specify which process on the sending machine is to get it.

The *UDP length* field includes the 8-byte header and the data. The *UDP checksum* is optional and stored as 0 if not computed (a true computed 0 is stored as all 1s). Turning it off is foolish unless the quality of the data does not matter (e.g., digitized speech).

It is probably worth mentioning explicitly some of the things that UDP does *not* do. It does not do flow control, error control, or retransmission upon receipt of a bad segment. All of that is up to the user processes. What it does do is provide an interface to the IP protocol with the added feature of demultiplexing multiple processes using the ports. That is all it does. For applications that need to have precise control over the packet flow, error control, or timing, UDP provides just what the doctor ordered.

One area where UDP is especially useful is in client-server situations. Often, the client sends a short request to the server and expects a short reply back. If either the request or reply is lost, the client can just time out and try again. Not only is the code simple, but fewer messages are required (one in each direction) than with a protocol requiring an initial setup.

An application that uses UDP this way is DNS (the Domain Name System), which we will study in Chap. 7. In brief, a program that needs to look up the IP address of some host name, for example, *www.cs.berkeley.edu*, can send a UDP packet containing the host name to a DNS server. The server replies with a UDP packet containing the host's IP address. No setup is needed in advance and no release is needed afterward. Just two messages go over the network.

6.4.2 Remote Procedure Call

In a certain sense, sending a message to a remote host and getting a reply back is a lot like making a function call in a programming language. In both cases you start with one or more parameters and you get back a result. This observation has led people to try to arrange request-reply interactions on networks to be cast in the

form of procedure calls. Such an arrangement makes network applications much easier to program and more familiar to deal with. For example, just imagine a procedure named *get_IP_address* (*host_name*) that works by sending a UDP packet to a DNS server and waiting for the reply, timing out and trying again if one is not forthcoming quickly enough. In this way, all the details of networking can be hidden from the programmer.

The key work in this area was done by Birrell and Nelson (1984). In a nutshell, what Birrell and Nelson suggested was allowing programs to call procedures located on remote hosts. When a process on machine 1 calls a procedure on machine 2, the calling process on 1 is suspended and execution of the called procedure takes place on 2. Information can be transported from the caller to the callee in the parameters and can come back in the procedure result. No message passing is visible to the programmer. This technique is known as **RPC** (**Remote Procedure Call**) and has become the basis for many networking applications. Traditionally, the calling procedure is known as the client and the called procedure is known as the server, and we will use those names here too.

The idea behind RPC is to make a remote procedure call look as much as possible like a local one. In the simplest form, to call a remote procedure, the client program must be bound with a small library procedure, called the **client stub**, that represents the server procedure in the client's address space. Similarly, the server is bound with a procedure called the **server stub**. These procedures hide the fact that the procedure call from the client to the server is not local.

The actual steps in making an RPC are shown in Fig. 6-24. Step 1 is the client calling the client stub. This call is a local procedure call, with the parameters pushed onto the stack in the normal way. Step 2 is the client stub packing the parameters into a message and making a system call to send the message. Packing the parameters is called **marshaling**. Step 3 is the kernel sending the message from the client machine to the server machine. Step 4 is the kernel passing the incoming packet to the server stub. Finally, step 5 is the server stub calling the server procedure with the unmarshaled parameters. The reply traces the same path in the other direction.

The key item to note here is that the client procedure, written by the user, just makes a normal (i.e., local) procedure call to the client stub, which has the same name as the server procedure. Since the client procedure and client stub are in the same address space, the parameters are passed in the usual way. Similarly, the server procedure is called by a procedure in its address space with the parameters it expects. To the server procedure, nothing is unusual. In this way, instead of I/O being done on sockets, network communication is done by faking a normal procedure call.

Despite the conceptual elegance of RPC, there are a few snakes hiding under the grass. A big one is the use of pointer parameters. Normally, passing a pointer to a procedure is not a problem. The called procedure can use the pointer in the same way the caller can because both procedures live in the same virtual address

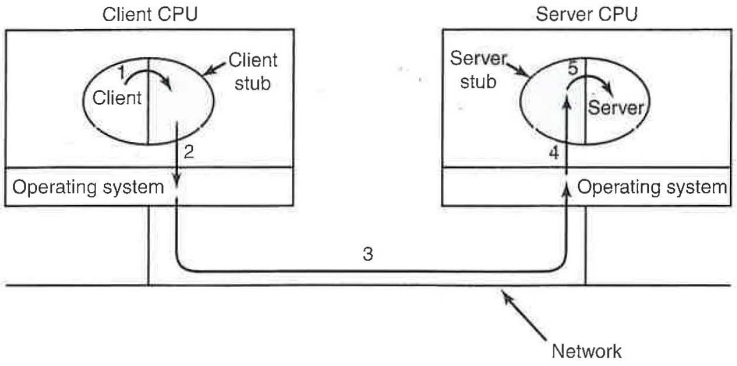

Figure 6-24. Steps in making a remote procedure call. The stubs are shaded.

space. With RPC, passing pointers is impossible because the client and server are in different address spaces.

In some cases, tricks can be used to make it possible to pass pointers. Suppose that the first parameter is a pointer to an integer, *k*. The client stub can marshal *k* and send it along to the server. The server stub then creates a pointer to *k* and passes it to the server procedure, just as it expects. When the server procedure returns control to the server stub, the latter sends *k* back to the client where the new *k* is copied over the old one, just in case the server changed it. In effect, the standard calling sequence of call-by-reference has been replaced by copy-restore. Unfortunately, this trick does not always work, for example, if the pointer points to a graph or other complex data structure. For this reason, some restrictions must be placed on parameters to procedures called remotely.

A second problem is that in weakly-typed languages, like C, it is perfectly legal to write a procedure that computes the inner product of two vectors (arrays), without specifying how large either one is. Each could be terminated by a special value known only to the calling and called procedure. Under these circumstances, it is essentially impossible for the client stub to marshal the parameters: it has no way of determining how large they are.

A third problem is that it is not always possible to deduce the types of the parameters, not even from a formal specification or the code itself. An example is *printf*, which may have any number of parameters (at least one), and the parameters can be an arbitrary mixture of integers, shorts, longs, characters, strings, floating-point numbers of various lengths, and other types. Trying to call *printf* as a remote procedure would be practically impossible because C is so permissive. However, a rule saying that RPC can be used provided that you do not program in C (or C++) would not be popular.

A fourth problem relates to the use of global variables. Normally, the calling and called procedure can communicate by using global variables, in addition to

communicating via parameters. If the called procedure is now moved to a remote machine, the code will fail because the global variables are no longer shared.

These problems are not meant to suggest that RPC is hopeless. In fact, it is widely used, but some restrictions are needed to make it work well in practice.

Of course, RPC need not use UDP packets, but RPC and UDP are a good fit and UDP is commonly used for RPC. However, when the parameters or results may be larger than the maximum UDP packet or when the operation requested is not idempotent (i.e., cannot be repeated safely, such as when incrementing a counter), it may be necessary to set up a TCP connection and send the request over it rather than use UDP.

6.4.3 The Real-Time Transport Protocol

Client-server RPC is one area in which UDP is widely used. Another one is real-time multimedia applications. In particular, as Internet radio, Internet telephony, music-on-demand, videoconferencing, video-on-demand, and other multimedia applications became more commonplace, people discovered that each application was reinventing more or less the same real-time transport protocol. It gradually became clear that having a generic real-time transport protocol for multiple applications would be a good idea. Thus was **RTP** (**Real-time Transport Protocol**) born. It is described in RFC 1889 and is now in widespread use.

The position of RTP in the protocol stack is somewhat strange. It was decided to put RTP in user space and have it (normally) run over UDP. It operates as follows. The multimedia application consists of multiple audio, video, text, and possibly other streams. These are fed into the RTP library, which is in user space along with the application. This library then multiplexes the streams and encodes them in RTP packets, which it then stuffs into a socket. At the other end of the socket (in the operating system kernel), UDP packets are generated and embedded in IP packets. If the computer is on an Ethernet, the IP packets are then put in Ethernet frames for transmission. The protocol stack for this situation is shown in Fig. 6-25(a). The packet nesting is shown in Fig. 6-25(b).

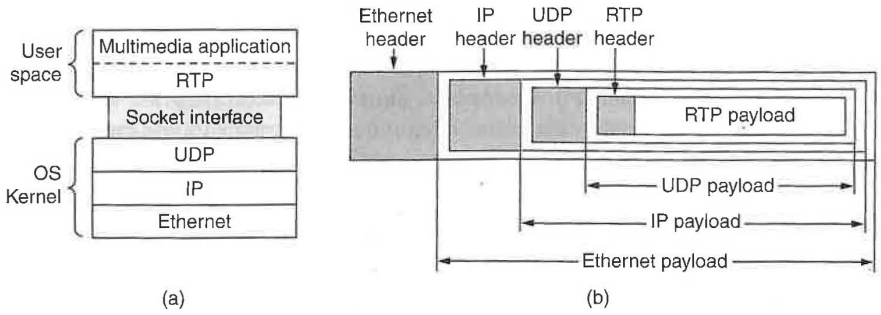

Figure 6-25. (a) The position of RTP in the protocol stack. (b) Packet nesting.

As a consequence of this design, it is a little hard to say which layer RTP is in. Since it runs in user space and is linked to the application program, it certainly looks like an application protocol. On the other hand, it is a generic, application-independent protocol that just provides transport facilities, so it also looks like a transport protocol. Probably the best description is that it is a transport protocol that is implemented in the application layer.

The basic function of RTP is to multiplex several real-time data streams onto a single stream of UDP packets. The UDP stream can be sent to a single destination (unicasting) or to multiple destinations (multicasting). Because RTP just uses normal UDP, its packets are not treated specially by the routers unless some normal IP quality-of-service features are enabled. In particular, there are no special guarantees about delivery, jitter, etc.

Each packet sent in an RTP stream is given a number one higher than its predecessor. This numbering allows the destination to determine if any packets are missing. If a packet is missing, the best action for the destination to take is to approximate the missing value by interpolation. Retransmission is not a practical option since the retransmitted packet would probably arrive too late to be useful. As a consequence, RTP has no flow control, no error control, no acknowledgements, and no mechanism to request retransmissions.

Each RTP payload may contain multiple samples, and they may be coded any way that the application wants. To allow for interworking, RTP defines several profiles (e.g., a single audio stream), and for each profile, multiple encoding formats may be allowed. For example, a single audio stream may be encoded as 8-bit PCM samples at 8 kHz, delta encoding, predictive encoding, GSM encoding, MP3, and so on. RTP provides a header field in which the source can specify the encoding but is otherwise not involved in how encoding is done.

Another facility many real-time applications need is timestamping. The idea here is to allow the source to associate a timestamp with the first sample in each packet. The timestamps are relative to the start of the stream, so only the differences between timestamps are significant. The absolute values have no meaning. This mechanism allows the destination to do a small amount of buffering and play each sample the right number of milliseconds after the start of the stream, independently of when the packet containing the sample arrived. Not only does timestamping reduce the effects of jitter, but it also allows multiple streams to be synchronized with each other. For example, a digital television program might have a video stream and two audio streams. The two audio streams could be for stereo broadcasts or for handling films with an original language soundtrack and a soundtrack dubbed into the local language, giving the viewer a choice. Each stream comes from a different physical device, but if they are timestamped from a single counter, they can be played back synchronously, even if the streams are transmitted somewhat erratically.

The RTP header is illustrated in Fig. 6-26. It consists of three 32-bit words and potentially some extensions. The first word contains the *Version* field, which

is already at 2. Let us hope this version is very close to the ultimate version since there is only one code point left (although 3 could be defined as meaning that the real version was in an extension word).

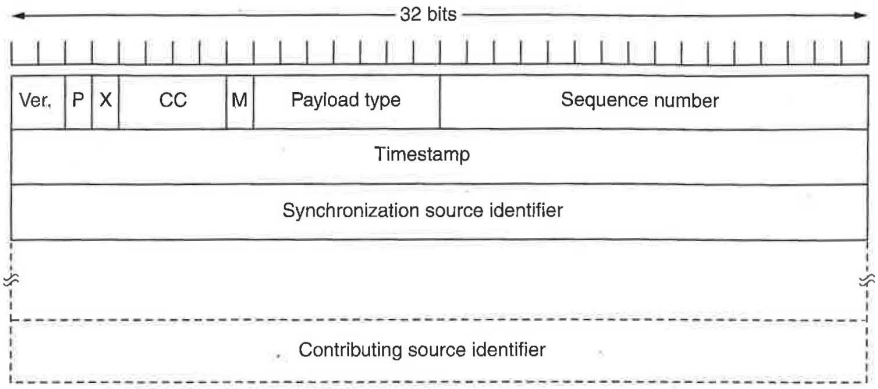

Figure 6-26. The RTP header.

The *P* bit indicates that the packet has been padded to a multiple of 4 bytes. The last padding byte tells how many bytes were added. The *X* bit indicates that an extension header is present. The format and meaning of the extension header are not defined. The only thing that is defined is that the first word of the extension gives the length. This is an escape hatch for any unforeseen requirements.

The *CC* field tells how many contributing sources are present, from 0 to 15 (see below). The *M* bit is an application-specific marker bit. It can be used to mark the start of a video frame, the start of a word in an audio channel, or something else that the application understands. The *Payload type* field tells which encoding algorithm has been used (e.g., uncompressed 8-bit audio, MP3, etc.). Since every packet carries this field, the encoding can change during transmission. The *Sequence number* is just a counter that is incremented on each RTP packet sent. It is used to detect lost packets.

The timestamp is produced by the stream's source to note when the first sample in the packet was made. This value can help reduce jitter at the receiver by decoupling the playback from the packet arrival time. The *Synchronization source identifier* tells which stream the packet belongs to. It is the method used to multiplex and demultiplex multiple data streams onto a single stream of UDP packets. Finally, the *Contributing source identifiers*, if any, are used when mixers are present in the studio. In that case, the mixer is the synchronizing source, and the streams being mixed are listed here.

RTP has a little sister protocol (little sibling protocol?) called **RTCP** (**Real-time Transport Control Protocol**). It handles feedback, synchronization, and the user interface but does not transport any data. The first function can be used

to provide feedback on delay, jitter, bandwidth, congestion, and other network properties to the sources. This information can be used by the encoding process to increase the data rate (and give better quality) when the network is functioning well and to cut back the data rate when there is trouble in the network. By providing continuous feedback, the encoding algorithms can be continuously adapted to provide the best quality possible under the current circumstances. For example, if the bandwidth increases or decreases during the transmission, the encoding may switch from MP3 to 8-bit PCM to delta encoding as required. The *Payload type* field is used to tell the destination what encoding algorithm is used for the current packet, making it possible to vary it on demand.

RTCP also handles interstream synchronization. The problem is that different streams may use different clocks, with different granularities and different drift rates. RTCP can be used to keep them in sync.

Finally, RTCP provides a way for naming the various sources (e.g., in ASCII text). This information can be displayed on the receiver's screen to indicate who is talking at the moment.

More information about RTP can be found in (Perkins, 2002).

6.5 THE INTERNET TRANSPORT PROTOCOLS: TCP

UDP is a simple protocol and it has some niche uses, such as client-server interactions and multimedia, but for most Internet applications, reliable, sequenced delivery is needed. UDP cannot provide this, so another protocol is required. It is called TCP and is the main workhorse of the Internet. Let us now study it in detail.

6.5.1 Introduction to TCP

TCP (**Transmission Control Protocol**) was specifically designed to provide a reliable end-to-end byte stream over an unreliable internetwork. An internetwork differs from a single network because different parts may have wildly different topologies, bandwidths, delays, packet sizes, and other parameters. TCP was designed to dynamically adapt to properties of the internetwork and to be robust in the face of many kinds of failures.

TCP was formally defined in RFC 793. As time went on, various errors and inconsistencies were detected, and the requirements were changed in some areas. These clarifications and some bug fixes are detailed in RFC 1122. Extensions are given in RFC 1323.

Each machine supporting TCP has a TCP transport entity, either a library procedure, a user process, or part of the kernel. In all cases, it manages TCP streams and interfaces to the IP layer. A TCP entity accepts user data streams from local processes, breaks them up into pieces not exceeding 64 KB (in practice, often

1460 data bytes in order to fit in a single Ethernet frame with the IP and TCP headers), and sends each piece as a separate IP datagram. When datagrams containing TCP data arrive at a machine, they are given to the TCP entity, which reconstructs the original byte streams. For simplicity, we will sometimes use just "TCP" to mean the TCP transport entity (a piece of software) or the TCP protocol (a set of rules). From the context it will be clear which is meant. For example, in "The user gives TCP the data," the TCP transport entity is clearly intended.

The IP layer gives no guarantee that datagrams will be delivered properly, so it is up to TCP to time out and retransmit them as need be. Datagrams that do arrive may well do so in the wrong order; it is also up to TCP to reassemble them into messages in the proper sequence. In short, TCP must furnish the reliability that most users want and that IP does not provide.

6.5.2 The TCP Service Model

TCP service is obtained by both the sender and receiver creating end points, called sockets, as discussed in Sec. 6.1.3. Each socket has a socket number (address) consisting of the IP address of the host and a 16-bit number local to that host, called a **port**. A port is the TCP name for a TSAP. For TCP service to be obtained, a connection must be explicitly established between a socket on the sending machine and a socket on the receiving machine. The socket calls are listed in Fig. 6-5.

A socket may be used for multiple connections at the same time. In other words, two or more connections may terminate at the same socket. Connections are identified by the socket identifiers at both ends, that is, (*socket1*, *socket2*). No virtual circuit numbers or other identifiers are used.

Port numbers below 1024 are called **well-known ports** and are reserved for standard services. For example, any process wishing to establish a connection to a host to transfer a file using FTP can connect to the destination host's port 21 to contact its FTP daemon. The list of well-known ports is given at *www.iana.org*. Over 300 have been assigned. A few of the better known ones are listed in Fig. 6-27.

It would certainly be possible to have the FTP daemon attach itself to port 21 at boot time, the telnet daemon to attach itself to port 23 at boot time, and so on. However, doing so would clutter up memory with daemons that were idle most of the time. Instead, what is generally done is to have a single daemon, called **inetd** (**Internet daemon**) in UNIX, attach itself to multiple ports and wait for the first incoming connection. When that occurs, inetd forks off a new process and executes the appropriate daemon in it, letting that daemon handle the request. In this way, the daemons other than inetd are only active when there is work for them to do. Inetd learns which ports it is to use from a configuration file. Consequently, the system administrator can set up the system to have permanent daemons on the busiest ports (e.g., port 80) and inetd on the rest.

Port	Protocol	Use
21	FTP	File transfer
23	Telnet	Remote login
25	SMTP	E-mail
69	TFTP	Trivial file transfer protocol
79	Finger	Lookup information about a user
80	HTTP	World Wide Web
110	POP-3	Remote e-mail access
119	NNTP	USENET news

Figure 6-27. Some assigned ports.

All TCP connections are full duplex and point-to-point. Full duplex means that traffic can go in both directions at the same time. Point-to-point means that each connection has exactly two end points. TCP does not support multicasting or broadcasting.

A TCP connection is a byte stream, not a message stream. Message boundaries are not preserved end to end. For example, if the sending process does four 512-byte writes to a TCP stream, these data may be delivered to the receiving process as four 512-byte chunks, two 1024-byte chunks, one 2048-byte chunk (see Fig. 6-28), or some other way. There is no way for the receiver to detect the unit(s) in which the data were written.

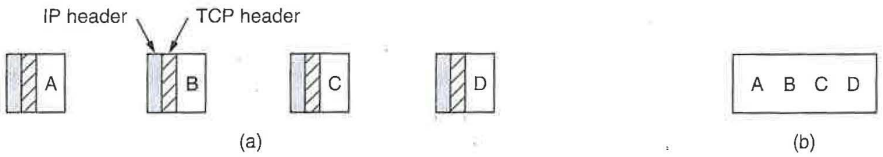

Figure 6-28. (a) Four 512-byte segments sent as separate IP datagrams. (b) The 2048 bytes of data delivered to the application in a single READ call.

Files in UNIX have this property too. The reader of a file cannot tell whether the file was written a block at a time, a byte at a time, or all in one blow. As with a UNIX file, the TCP software has no idea of what the bytes mean and no interest in finding out. A byte is just a byte.

When an application passes data to TCP, TCP may send it immediately or buffer it (in order to collect a larger amount to send at once), at its discretion. However, sometimes, the application really wants the data to be sent immediately. For example, suppose a user is logged in to a remote machine. After a command line has been finished and the carriage return typed, it is essential that the line be

shipped off to the remote machine immediately and not buffered until the next line comes in. To force data out, applications can use the PUSH flag, which tells TCP not to delay the transmission.

Some early applications used the PUSH flag as a kind of marker to delineate messages boundaries. While this trick sometimes works, it sometimes fails since not all implementations of TCP pass the PUSH flag to the application on the receiving side. Furthermore, if additional PUSHes come in before the first one has been transmitted (e.g., because the output line is busy), TCP is free to collect all the PUSHed data into a single IP datagram, with no separation between the various pieces.

One last feature of the TCP service that is worth mentioning here is **urgent data**. When an interactive user hits the DEL or CTRL-C key to break off a remote computation that has already begun, the sending application puts some control information in the data stream and gives it to TCP along with the URGENT flag. This event causes TCP to stop accumulating data and transmit everything it has for that connection immediately.

When the urgent data are received at the destination, the receiving application is interrupted (e.g., given a signal in UNIX terms) so it can stop whatever it was doing and read the data stream to find the urgent data. The end of the urgent data is marked so the application knows when it is over. The start of the urgent data is not marked. It is up to the application to figure that out. This scheme basically provides a crude signaling mechanism and leaves everything else up to the application.

6.5.3 The TCP Protocol

In this section we will give a general overview of the TCP protocol. In the next one we will go over the protocol header, field by field.

A key feature of TCP, and one which dominates the protocol design, is that every byte on a TCP connection has its own 32-bit sequence number. When the Internet began, the lines between routers were mostly 56-kbps leased lines, so a host blasting away at full speed took over 1 week to cycle through the sequence numbers. At modern network speeds, the sequence numbers can be consumed at an alarming rate, as we will see later. Separate 32-bit sequence numbers are used for acknowledgements and for the window mechanism, as discussed below.

The sending and receiving TCP entities exchange data in the form of segments. A **TCP segment** consists of a fixed 20-byte header (plus an optional part) followed by zero or more data bytes. The TCP software decides how big segments should be. It can accumulate data from several writes into one segment or can split data from one write over multiple segments. Two limits restrict the segment size. First, each segment, including the TCP header, must fit in the 65,515-byte IP payload. Second, each network has a **maximum transfer unit**, or **MTU**,

and each segment must fit in the MTU. In practice, the MTU is generally 1500 bytes (the Ethernet payload size) and thus defines the upper bound on segment size.

The basic protocol used by TCP entities is the sliding window protocol. When a sender transmits a segment, it also starts a timer. When the segment arrives at the destination, the receiving TCP entity sends back a segment (with data if any exist, otherwise without data) bearing an acknowledgement number equal to the next sequence number it expects to receive. If the sender's timer goes off before the acknowledgement is received, the sender transmits the segment again.

Although this protocol sounds simple, there are a number of sometimes subtle ins and outs, which we will cover below. Segments can arrive out of order, so bytes 3072–4095 can arrive but cannot be acknowledged because bytes 2048–3071 have not turned up yet. Segments can also be delayed so long in transit that the sender times out and retransmits them. The retransmissions may include different byte ranges than the original transmission, requiring a careful administration to keep track of which bytes have been correctly received so far. However, since each byte in the stream has its own unique offset, it can be done.

TCP must be prepared to deal with these problems and solve them in an efficient way. A considerable amount of effort has gone into optimizing the performance of TCP streams, even in the face of network problems. A number of the algorithms used by many TCP implementations will be discussed below.

6.5.4 The TCP Segment Header

Figure 6-29 shows the layout of a TCP segment. Every segment begins with a fixed-format, 20-byte header. The fixed header may be followed by header options. After the options, if any, up to $65,535 - 20 - 20 = 65,495$ data bytes may follow, where the first 20 refer to the IP header and the second to the TCP header. Segments without any data are legal and are commonly used for acknowledgements and control messages.

Let us dissect the TCP header field by field. The *Source port* and *Destination port* fields identify the local end points of the connection. The well-known ports are defined at *www.iana.org* but each host can allocate the others as it wishes. A port plus its host's IP address forms a 48-bit unique end point. The source and destination end points together identify the connection.

The *Sequence number* and *Acknowledgement number* fields perform their usual functions. Note that the latter specifies the next byte expected, not the last byte correctly received. Both are 32 bits long because every byte of data is numbered in a TCP stream.

The *TCP header length* tells how many 32-bit words are contained in the TCP header. This information is needed because the *Options* field is of variable length, so the header is, too. Technically, this field really indicates the start of the data

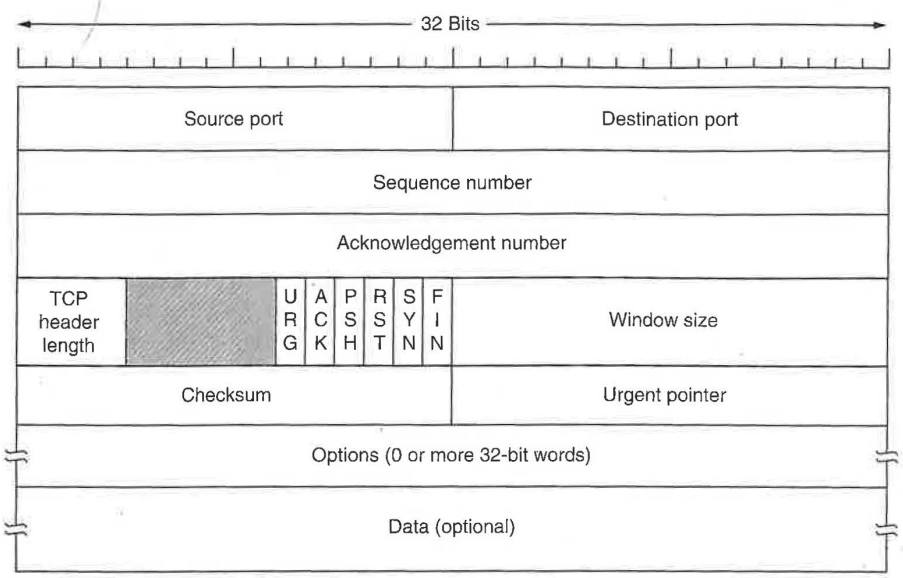

Figure 6-29. The TCP header.

within the segment, measured in 32-bit words, but that number is just the header length in words, so the effect is the same.

Next comes a 6-bit field that is not used. The fact that this field has survived intact for over a quarter of a century is testimony to how well thought out TCP is. Lesser protocols would have needed it to fix bugs in the original design.

Now come six 1-bit flags. *URG* is set to 1 if the *Urgent pointer* is in use. The *Urgent pointer* is used to indicate a byte offset from the current sequence number at which urgent data are to be found. This facility is in lieu of interrupt messages. As we mentioned above, this facility is a bare-bones way of allowing the sender to signal the receiver without getting TCP itself involved in the reason for the interrupt.

The *ACK* bit is set to 1 to indicate that the *Acknowledgement number* is valid. If *ACK* is 0, the segment does not contain an acknowledgement so the *Acknowledgement number* field is ignored.

The *PSH* bit indicates PUSHed data. The receiver is hereby kindly requested to deliver the data to the application upon arrival and not buffer it until a full buffer has been received (which it might otherwise do for efficiency).

The *RST* bit is used to reset a connection that has become confused due to a host crash or some other reason. It is also used to reject an invalid segment or refuse an attempt to open a connection. In general, if you get a segment with the *RST* bit on, you have a problem on your hands.

The *SYN* bit is used to establish connections. The connection request has *SYN* = 1 and *ACK* = 0 to indicate that the piggyback acknowledgement field is not in use. The connection reply does bear an acknowledgement, so it has *SYN* = 1 and *ACK* = 1. In essence the *SYN* bit is used to denote CONNECTION REQUEST and CONNECTION ACCEPTED, with the *ACK* bit used to distinguish between those two possibilities.

The *FIN* bit is used to release a connection. It specifies that the sender has no more data to transmit. However, after closing a connection, the closing process may continue to receive data indefinitely. Both *SYN* and *FIN* segments have sequence numbers and are thus guaranteed to be processed in the correct order.

Flow control in TCP is handled using a variable-sized sliding window. The *Window size* field tells how many bytes may be sent starting at the byte acknowledged. A *Window size* field of 0 is legal and says that the bytes up to and including *Acknowledgement number* − 1 have been received, but that the receiver is currently badly in need of a rest and would like no more data for the moment, thank you. The receiver can later grant permission to send by transmitting a segment with the same *Acknowledgement number* and a nonzero *Window size* field.

In the protocols of Chap. 3, acknowledgements of frames received and permission to send new frames were tied together. This was a consequence of a fixed window size for each protocol. In TCP, acknowledgements and permission to send additional data are completely decoupled. In effect, a receiver can say: I have received bytes up through k but I do not want any more just now. This decoupling (in fact, a variable-sized window) gives additional flexibility. We will study it in detail below.

A *Checksum* is also provided for extra reliability. It checksums the header, the data, and the conceptual pseudoheader shown in Fig. 6-30. When performing this computation, the TCP *Checksum* field is set to zero and the data field is padded out with an additional zero byte if its length is an odd number. The checksum algorithm is simply to add up all the 16-bit words in one's complement and then to take the one's complement of the sum. As a consequence, when the receiver performs the calculation on the entire segment, including the *Checksum* field, the result should be 0.

The pseudoheader contains the 32-bit IP addresses of the source and destination machines, the protocol number for TCP (6), and the byte count for the TCP segment (including the header). Including the pseudoheader in the TCP checksum computation helps detect misdelivered packets, but including it also violates the protocol hierarchy since the IP addresses in it belong to the IP layer, not to the TCP layer. UDP uses the same pseudoheader for its checksum.

The *Options* field provides a way to add extra facilities not covered by the regular header. The most important option is the one that allows each host to specify the maximum TCP payload it is willing to accept. Using large segments is more efficient than using small ones because the 20-byte header can then be amortized over more data, but small hosts may not be able to handle big segments.

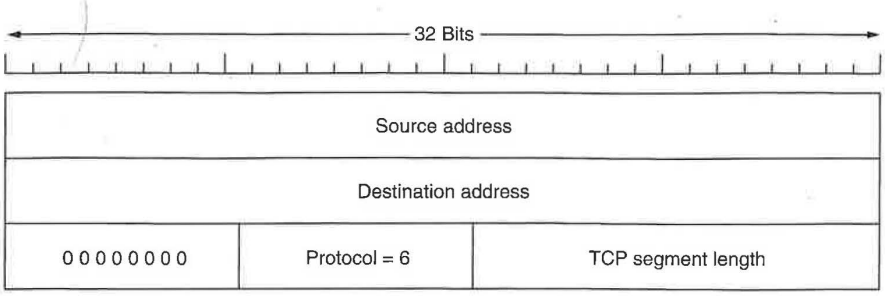

Figure 6-30. The pseudoheader included in the TCP checksum.

During connection setup, each side can announce its maximum and see its partner's. If a host does not use this option, it defaults to a 536-byte payload. All Internet hosts are required to accept TCP segments of 536 + 20 = 556 bytes. The maximum segment size in the two directions need not be the same.

For lines with high bandwidth, high delay, or both, the 64-KB window is often a problem. On a T3 line (44.736 Mbps), it takes only 12 msec to output a full 64-KB window. If the round-trip propagation delay is 50 msec (which is typical for a transcontinental fiber), the sender will be idle 3/4 of the time waiting for acknowledgements. On a satellite connection, the situation is even worse. A larger window size would allow the sender to keep pumping data out, but using the 16-bit *Window size* field, there is no way to express such a size. In RFC 1323, a *Window scale* option was proposed, allowing the sender and receiver to negotiate a window scale factor. This number allows both sides to shift the *Window size* field up to 14 bits to the left, thus allowing windows of up to 2^{30} bytes. Most TCP implementations now support this option.

Another option proposed by RFC 1106 and now widely implemented is the use of the selective repeat instead of go back n protocol. If the receiver gets one bad segment and then a large number of good ones, the normal TCP protocol will eventually time out and retransmit all the unacknowledged segments, including all those that were received correctly (i.e., the go back n protocol). RFC 1106 introduced NAKs to allow the receiver to ask for a specific segment (or segments). After it gets these, it can acknowledge all the buffered data, thus reducing the amount of data retransmitted.

6.5.5 TCP Connection Establishment

Connections are established in TCP by means of the three-way handshake discussed in Sec. 6.2.2. To establish a connection, one side, say, the server, passively waits for an incoming connection by executing the LISTEN and ACCEPT primitives, either specifying a specific source or nobody in particular.

The other side, say, the client, executes a CONNECT primitive, specifying the IP address and port to which it wants to connect, the maximum TCP segment size it is willing to accept, and optionally some user data (e.g., a password). The CONNECT primitive sends a TCP segment with the *SYN* bit on and *ACK* bit off and waits for a response.

When this segment arrives at the destination, the TCP entity there checks to see if there is a process that has done a LISTEN on the port given in the *Destination port* field. If not, it sends a reply with the *RST* bit on to reject the connection.

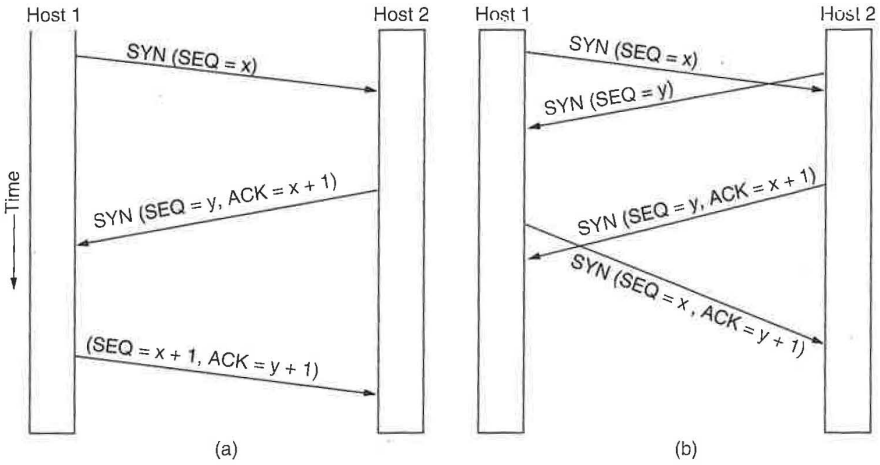

Figure 6-31. (a) TCP connection establishment in the normal case. (b) Call collision.

If some process is listening to the port, that process is given the incoming TCP segment. It can then either accept or reject the connection. If it accepts, an acknowledgement segment is sent back. The sequence of TCP segments sent in the normal case is shown in Fig. 6-31(a). Note that a *SYN* segment consumes 1 byte of sequence space so that it can be acknowledged unambiguously.

In the event that two hosts simultaneously attempt to establish a connection between the same two sockets, the sequence of events is as illustrated in Fig. 6-31(b). The result of these events is that just one connection is established, not two because connections are identified by their end points. If the first setup results in a connection identified by (x, y) and the second one does too, only one table entry is made, namely, for (x, y).

The initial sequence number on a connection is not 0 for the reasons we discussed earlier. A clock-based scheme is used, with a clock tick every 4 μsec. For additional safety, when a host crashes, it may not reboot for the maximum packet lifetime to make sure that no packets from previous connections are still roaming around the Internet somewhere.

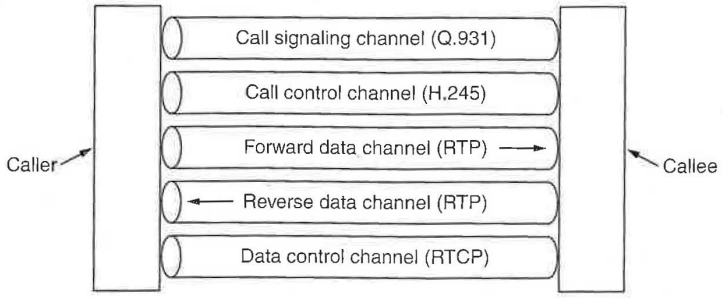

Figure 7-66. Logical channels between the caller and callee during a call.

SIP—The Session Initiation Protocol

H.323 was designed by ITU. Many people in the Internet community saw it as a typical telco product: large, complex, and inflexible. Consequently, IETF set up a committee to design a simpler and more modular way to do voice over IP. The major result to date is the **SIP** (**Session Initiation Protocol**), which is described in RFC 3261. This protocol describes how to set up Internet telephone calls, video conferences, and other multimedia connections. Unlike H.323, which is a complete protocol suite, SIP is a single module, but it has been designed to interwork well with existing Internet applications. For example, it defines telephone numbers as URLs, so that Web pages can contain them, allowing a click on a link to initiate a telephone call (the same way the *mailto* scheme allows a click on a link to bring up a program to send an e-mail message).

SIP can establish two-party sessions (ordinary telephone calls), multiparty sessions (where everyone can hear and speak), and multicast sessions (one sender, many receivers). The sessions may contain audio, video, or data, the latter being useful for multiplayer real-time games, for example. SIP just handles setup, management, and termination of sessions. Other protocols, such as RTP/RTCP, are used for data transport. SIP is an application-layer protocol and can run over UDP or TCP.

SIP supports a variety of services, including locating the callee (who may not be at his home machine) and determining the callee's capabilities, as well as handling the mechanics of call setup and termination. In the simplest case, SIP sets up a session from the caller's computer to the callee's computer, so we will examine that case first.

Telephone numbers in SIP are represented as URLs using the *sip* scheme, for example, *sip:ilse@cs.university.edu* for a user named Ilse at the host specified by the DNS name *cs.university.edu*. SIP URLs may also contain IPv4 addresses, IPv6 address, or actual telephone numbers.

The SIP protocol is a text-based protocol modeled on HTTP. One party sends a message in ASCII text consisting of a method name on the first line, followed by additional lines containing headers for passing parameters. Many of the headers are taken from MIME to allow SIP to interwork with existing Internet applications. The six methods defined by the core specification are listed in Fig. 7-67.

Method	Description
INVITE	Request initiation of a session
ACK	Confirm that a session has been initiated
BYE	Request termination of a session
OPTIONS	Query a host about its capabilities
CANCEL	Cancel a pending request
REGISTER	Inform a redirection server about the user's current location

Figure 7-67. The SIP methods defined in the core specification.

To establish a session, the caller either creates a TCP connection with the callee and sends an *INVITE* message over it or sends the *INVITE* message in a UDP packet. In both cases, the headers on the second and subsequent lines describe the structure of the message body, which contains the caller's capabilities, media types, and formats. If the callee accepts the call, it responds with an HTTP-type reply code (a three-digit number using the groups of Fig. 7-42, 200 for acceptance). Following the reply-code line, the callee also may supply information about its capabilities, media types, and formats.

Connection is done using a three-way handshake, so the caller responds with an *ACK* message to finish the protocol and confirm receipt of the 200 message.

Either party may request termination of a session by sending a message containing the *BYE* method. When the other side acknowledges it, the session is terminated.

The *OPTIONS* method is used to query a machine about its own capabilities. It is typically used before a session is initiated to find out if that machine is even capable of voice over IP or whatever type of session is being contemplated.

The *REGISTER* method relates to SIP's ability to track down and connect to a user who is away from home. This message is sent to a SIP location server that keeps track of who is where. That server can later be queried to find the user's current location. The operation of redirection is illustrated in Fig. 7-68. Here the caller sends the *INVITE* message to a proxy server to hide the possible redirection. The proxy then looks up where the user is and sends the *INVITE* message there. It then acts as a relay for the subsequent messages in the three-way handshake. The *LOOKUP* and *REPLY* messages are not part of SIP; any convenient protocol can be used, depending on what kind of location server is used.

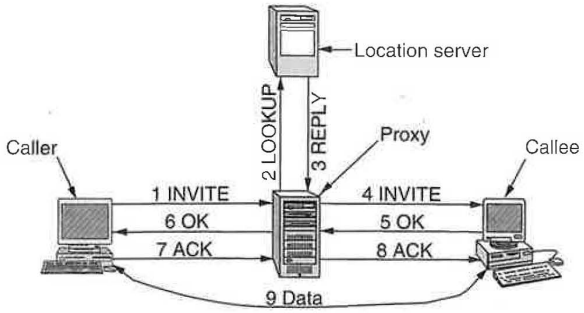

Figure 7-68. Use a proxy and redirection servers with SIP.

SIP has a variety of other features that we will not describe here, including call waiting, call screening, encryption, and authentication. It also has the ability to place calls from a computer to an ordinary telephone, if a suitable gateway between the Internet and telephone system is available.

Comparison of H.323 and SIP

H.323 and SIP have many similarities but also some differences. Both allow two-party and multiparty calls using both computers and telephones as end points. Both support parameter negotiation, encryption, and the RTP/RTCP protocols. A summary of the similarities and differences is given in Fig. 7-69.

Although the feature sets are similar, the two protocols differ widely in philosophy. H.323 is a typical, heavyweight, telephone-industry standard, specifying the complete protocol stack and defining precisely what is allowed and what is forbidden. This approach leads to very well defined protocols in each layer, easing the task of interoperability. The price paid is a large, complex, and rigid standard that is difficult to adapt to future applications.

In contrast, SIP is a typical Internet protocol that works by exchanging short lines of ASCII text. It is a lightweight module that interworks well with other Internet protocols but less well with existing telephone system signaling protocols. Because the IETF model of voice over IP is highly modular, it is flexible and can be adapted to new applications easily. The downside is potential interoperability problems, although these are addressed by frequent meetings where different implementers get together to test their systems.

Voice over IP is an up-and-coming topic. Consequently, there are several books on the subject already. A few examples are (Collins, 2001; Davidson and Peters, 2000; Kumar et al., 2001; and Wright, 2001). The May/June 2002 issue of *Internet Computing* has several articles on this topic.

Item	H.323	SIP
Designed by	ITU	IETF
Compatibility with PSTN	Yes	Largely
Compatibility with Internet	No	Yes
Architecture	Monolithic	Modular
Completeness	Full protocol stack	SIP just handles setup
Parameter negotiation	Yes	Yes
Call signaling	Q.931 over TCP	SIP over TCP or UDP
Message format	Binary	ASCII
Media transport	RTP/RTCP	RTP/RTCP
Multiparty calls	Yes	Yes
Multimedia conferences	Yes	No
Addressing	Host or telephone number	URL
Call termination	Explicit or TCP release	Explicit or timeout
Instant messaging	No	Yes
Encryption	Yes	Yes
Size of standards	1400 pages	250 pages
Implementation	Large and complex	Moderate
Status	Widely deployed	Up and coming

Figure 7-69. Comparison of H.323 and SIP

7.4.6 Introduction to Video

We have discussed the ear at length now; time to move on to the eye (no, this section is not followed by one on the nose). The human eye has the property that when an image appears on the retina, the image is retained for some number of milliseconds before decaying. If a sequence of images is drawn line by line at 50 images/sec, the eye does not notice that it is looking at discrete images. All video (i.e., television) systems exploit this principle to produce moving pictures.

Analog Systems

To understand video, it is best to start with simple, old-fashioned black-and-white television. To represent the two-dimensional image in front of it as a one-dimensional voltage as a function of time, the camera scans an electron beam rapidly across the image and slowly down it, recording the light intensity as it goes. At the end of the scan, called a **frame**, the beam retraces. This intensity as a function of time is broadcast, and receivers repeat the scanning process to re-

X

Z