timestamp clock continues to increase at a constant rate even if the source is inactive.

- *Synchronization source identifier (SSRC).* The SSRC field is 32 bits long. It identifies the source of the RTP stream. Typically, each stream in an RTP session has a distinct SSRC. The SSRC is not the IP address of the sender, but instead is a number that the source assigns randomly when the new stream is started. The probability that two streams get assigned the same SSRC is very small. Should this happen, the two sources pick a new SSRC value.

7.4.2 SIP

The Session Initiation Protocol (SIP), defined in [RFC 3261; RFC 5411], is an open and lightweight protocol that does the following:

- It provides mechanisms for establishing calls between a caller and a callee over an IP network. It allows the caller to notify the callee that it wants to start a call. It allows the participants to agree on media encodings. It also allows participants to end calls.
- It provides mechanisms for the caller to determine the current IP address of the callee. Users do not have a single, fixed IP address because they may be assigned addresses dynamically (using DHCP) and because they may have multiple IP devices, each with a different IP address.
- It provides mechanisms for call management, such as adding new media streams during the call, changing the encoding during the call, inviting new participants during the call, call transfer, and call holding.

Setting Up a Call to a Known IP Address

To understand the essence of SIP, it is best to take a look at a concrete example. In this example, Alice is at her PC and she wants to call Bob, who is also working at his PC. Alice's and Bob's PCs are both equipped with SIP-based software for making and receiving phone calls. In this initial example, we'll assume that Alice knows the IP address of Bob's PC. Figure 7.12 illustrates the SIP call-establishment process.

In Figure 7.12, we see that an SIP session begins when Alice sends Bob an INVITE message, which resembles an HTTP request message. This INVITE message is sent over UDP to the well-known port 5060 for SIP. (SIP messages can also be sent over TCP.) The INVITE message includes an identifier for Bob (bob@193.64.210.89), an indication of Alice's current IP address, an indication that Alice desires to receive audio, which is to be encoded in format AVP 0 (PCM encoded μ-law) and encapsulated in RTP, and an indication that she wants to receive

the RTP packets on port 38060. After receiving Alice's INVITE message, Bob sends an SIP response message, which resembles an HTTP response message. This response SIP message is also sent to the SIP port 5060. Bob's response includes a 200 OK as well as an indication of his IP address, his desired encoding and packetization for reception, and his port number to which the audio packets should be sent. Note that in this example Alice and Bob are going to use different audio-encoding mechanisms: Alice is asked to encode her audio with GSM whereas Bob is asked to encode his audio with PCM μ-law. After receiving Bob's response, Alice sends Bob an SIP acknowledgment message. After this SIP transaction, Bob and Alice can talk. (For visual convenience, Figure 7.12 shows Alice talking after Bob, but in truth they

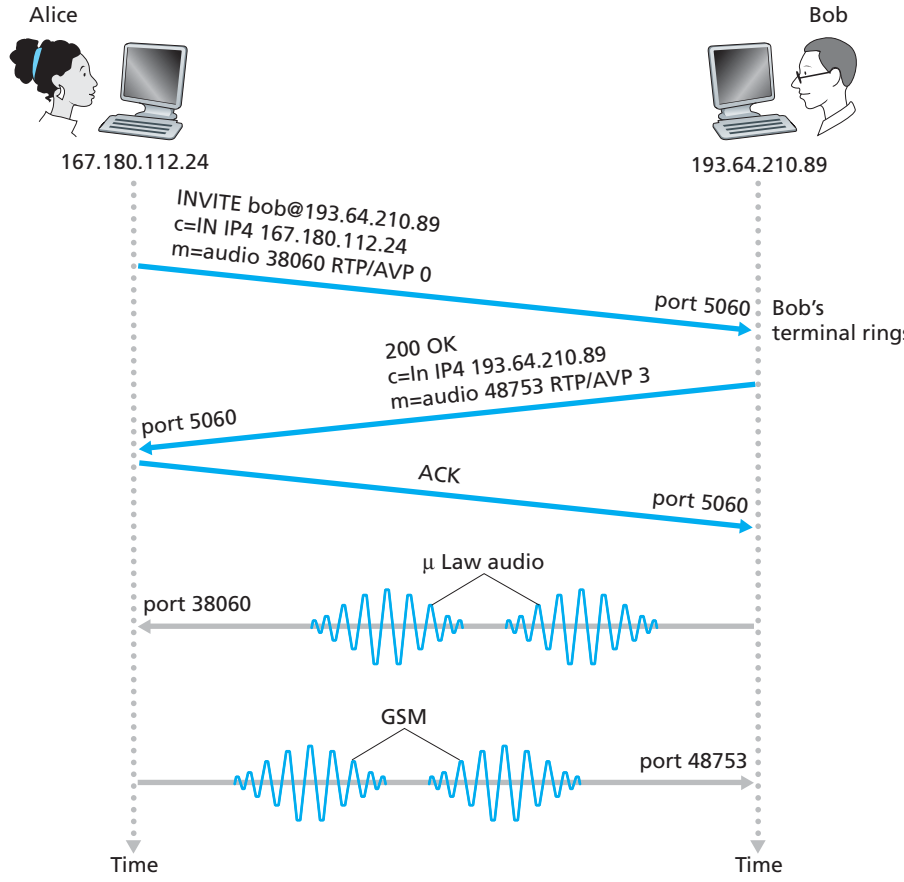

Figure 7.12 ♦ SIP call establishment when Alice knows Bob's IP address

would normally talk at the same time.) Bob will encode and packetize the audio as requested and send the audio packets to port number 38060 at IP address 167.180.112.24. Alice will also encode and packetize the audio as requested and send the audio packets to port number 48753 at IP address 193.64.210.89.

From this simple example, we have learned a number of key characteristics of SIP. First, SIP is an out-of-band protocol: The SIP messages are sent and received in sockets that are different from those used for sending and receiving the media data. Second, the SIP messages themselves are ASCII-readable and resemble HTTP messages. Third, SIP requires all messages to be acknowledged, so it can run over UDP or TCP.

In this example, let's consider what would happen if Bob does not have a PCM μ-law codec for encoding audio. In this case, instead of responding with 200 OK, Bob would likely respond with a 600 Not Acceptable and list in the message all the codecs he can use. Alice would then choose one of the listed codecs and send another INVITE message, this time advertising the chosen codec. Bob could also simply reject the call by sending one of many possible rejection reply codes. (There are many such codes, including "busy," "gone," "payment required," and "forbidden.")

SIP Addresses

In the previous example, Bob's SIP address is sip:bob@193.64.210.89. However, we expect many—if not most—SIP addresses to resemble e-mail addresses. For example, Bob's address might be sip:bob@domain.com. When Alice's SIP device sends an INVITE message, the message would include this e-mail-like address; the SIP infrastructure would then route the message to the IP device that Bob is currently using (as we'll discuss below). Other possible forms for the SIP address could be Bob's legacy phone number or simply Bob's first/middle/last name (assuming it is unique).

An interesting feature of SIP addresses is that they can be included in Web pages, just as people's e-mail addresses are included in Web pages with the mailto URL. For example, suppose Bob has a personal homepage, and he wants to provide a means for visitors to the homepage to call him. He could then simply include the URL sip:bob@domain.com. When the visitor clicks on the URL, the SIP application in the visitor's device is launched and an INVITE message is sent to Bob.

SIP Messages

In this short introduction to SIP, we'll not cover all SIP message types and headers. Instead, we'll take a brief look at the SIP INVITE message, along with a few common header lines. Let us again suppose that Alice wants to initiate a VoIP call to Bob, and this time Alice knows only Bob's SIP address, bob@domain.com, and

does not know the IP address of the device that Bob is currently using. Then her message might look something like this:

```
INVITE sip:bob@domain.com SIP/2.0
Via: SIP/2.0/UDP 167.180.112.24
From: sip:alice@hereway.com
To: sip:bob@domain.com
Call-ID: a2e3a@pigeon.hereway.com
Content-Type: application/sdp
Content-Length: 885

c=IN IP4 167.180.112.24
m=audio 38060 RTP/AVP 0
```

The INVITE line includes the SIP version, as does an HTTP request message. Whenever an SIP message passes through an SIP device (including the device that originates the message), it attaches a Via header, which indicates the IP address of the device. (We'll see soon that the typical INVITE message passes through many SIP devices before reaching the callee's SIP application.) Similar to an e-mail message, the SIP message includes a From header line and a To header line. The message includes a Call-ID, which uniquely identifies the call (similar to the message-ID in e-mail). It includes a Content-Type header line, which defines the format used to describe the content contained in the SIP message. It also includes a Content-Length header line, which provides the length in bytes of the content in the message. Finally, after a carriage return and line feed, the message contains the content. In this case, the content provides information about Alice's IP address and how Alice wants to receive the audio.

Name Translation and User Location

In the example in Figure 7.12, we assumed that Alice's SIP device knew the IP address where Bob could be contacted. But this assumption is quite unrealistic, not only because IP addresses are often dynamically assigned with DHCP, but also because Bob may have multiple IP devices (for example, different devices for his home, work, and car). So now let us suppose that Alice knows only Bob's e-mail address, bob@domain.com, and that this same address is used for SIP-based calls. In this case, Alice needs to obtain the IP address of the device that the user bob@domain.com is currently using. To find this out, Alice creates an INVITE message that begins with INVITE bob@domain.com SIP/2.0 and sends this message to an **SIP proxy**. The proxy will respond with an SIP reply that might include the IP address of the device that bob@domain.com is currently using. Alternatively, the reply might include the IP address of Bob's voicemail box, or it might include a URL of a Web page (that says "Bob is sleeping. Leave me alone!"). Also, the result returned by the proxy might depend on the caller: If the call is from Bob's wife, he

might accept the call and supply his IP address; if the call is from Bob's mother-in-law, he might respond with the URL that points to the I-am-sleeping Web page!

Now, you are probably wondering, how can the proxy server determine the current IP address for bob@domain.com? To answer this question, we need to say a few words about another SIP device, the **SIP registrar**. Every SIP user has an associated registrar. Whenever a user launches an SIP application on a device, the application sends an SIP register message to the registrar, informing the registrar of its current IP address. For example, when Bob launches his SIP application on his PDA, the application would send a message along the lines of:

```
REGISTER sip:domain.com SIP/2.0
Via: SIP/2.0/UDP 193.64.210.89
From: sip:bob@domain.com
To: sip:bob@domain.com
Expires: 3600
```

Bob's registrar keeps track of Bob's current IP address. Whenever Bob switches to a new SIP device, the new device sends a new register message, indicating the new IP address. Also, if Bob remains at the same device for an extended period of time, the device will send refresh register messages, indicating that the most recently sent IP address is still valid. (In the example above, refresh messages need to be sent every 3600 seconds to maintain the address at the registrar server.) It is worth noting that the registrar is analogous to a DNS authoritative name server: The DNS server translates fixed host names to fixed IP addresses; the SIP registrar translates fixed human identifiers (for example, bob@domain.com) to dynamic IP addresses. Often SIP registrars and SIP proxies are run on the same host.

Now let's examine how Alice's SIP proxy server obtains Bob's current IP address. From the preceding discussion we see that the proxy server simply needs to forward Alice's INVITE message to Bob's registrar/proxy. The registrar/proxy could then forward the message to Bob's current SIP device. Finally, Bob, having now received Alice's INVITE message, could send an SIP response to Alice.

As an example, consider Figure 7.13, in which jim@umass.edu, currently working on 217.123.56.89, wants to initiate a Voice-over-IP (VoIP) session with keith@upenn.edu, currently working on 197.87.54.21. The following steps are taken: (1) Jim sends an INVITE message to the umass SIP proxy. (2) The proxy does a DNS lookup on the SIP registrar upenn.edu (not shown in diagram) and then forwards the message to the registrar server. (3) Because keith@upenn.edu is no longer registered at the upenn registrar, the upenn registrar sends a redirect response, indicating that it should try keith@eurecom.fr. (4) The umass proxy sends an INVITE message to the eurecom SIP registrar. (5) The eurecom registrar knows the IP address of keith@eurecom.fr and forwards the INVITE message to the host 197.87.54.21, which is running Keith's SIP client. (6–8) An SIP response is sent back through registrars/proxies to the SIP client on 217.123.56.89. (9) Media is sent

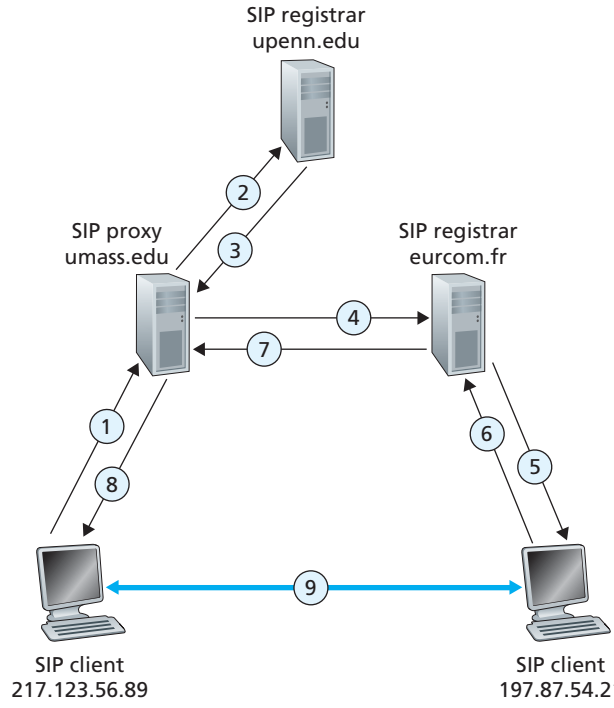

Figure 7.13 ♦ Session initiation, involving SIP proxies and registrars

directly between the two clients. (There is also an SIP acknowledgment message, which is not shown.)

Our discussion of SIP has focused on call initiation for voice calls. SIP, being a signaling protocol for initiating and ending calls in general, can be used for video conference calls as well as for text-based sessions. In fact, SIP has become a fundamental component in many instant messaging applications. Readers desiring to learn more about SIP are encouraged to visit Henning Schulzrinne's SIP Web site [Schulzrinne-SIP 2012]. In particular, on this site you will find open source software for SIP clients and servers [SIP Software 2012].

7.5 Network Support for Multimedia

In Sections 7.2 through 7.4, we learned how application-level mechanisms such as client buffering, prefetching, adapting media quality to available bandwidth, adaptive playout, and loss mitigation techniques can be used by multimedia applications

to improve a multimedia application's performance. We also learned how content distribution networks and P2P overlay networks can be used to provide a *system-level* approach for delivering multimedia content. These techniques and approaches are all designed to be used in today's best-effort Internet. Indeed, they are in use today precisely because the Internet provides only a single, best-effort class of service. But as designers of computer networks, we can't help but ask whether the *network* (rather than the applications or application-level infrastructure alone) might provide mechanisms to support multimedia content delivery. As we'll see shortly, the answer is, of course, "yes"! But we'll also see that a number of these new network-level mechanisms have yet to be widely deployed. This may be due to their complexity and to the fact that application-level techniques together with best-effort service and properly dimensioned network resources (for example, bandwidth) can indeed provide a "good-enough" (even if not-always-perfect) end-to-end multimedia delivery service.

Table 7.4 summarizes three broad approaches towards providing network-level support for multimedia applications.

- *Making the best of best-effort service.* The application-level mechanisms and infrastructure that we studied in Sections 7.2 through 7.4 can be successfully used in a well-dimensioned network where packet loss and excessive end-to-end

Approach	Granularity	Guarantee	Mechanisms	Complexity	Deployment to date
Making the best of best-effort service.	all traffic treated equally	none, or soft	application-layer support, CDNs, overlays, network-level resource provisioning	minimal	everywhere
Differentiated service	different classes of traffic treated differently	none, or soft	packet marking, policing, scheduling	medium	some
Per-connection Quality-of-Service (QoS) Guarantees	each source-destination flows treated differently	soft or hard, once flow is admitted	packet marking, policing, scheduling; call admission and signaling	light	little

Table 7.4 ♦ Three network-level approaches to supporting multimedia applications