CHAPTER 6

Wireless and Mobile Networks

In the telephony world, the past 15 years have arguably been the golden years of cellular telephony. The number of worldwide mobile cellular subscribers increased from 34 million in 1993 to nearly 5.5 billion subscribers by 2011, with the number of cellular subscribers now surpassing the number of wired telephone lines. The many advantages of cell phones are evident to all—anywhere, anytime, untethered access to the global telephone network via a highly portable lightweight device. With the advent of laptops, palmtops, smartphones, and their promise of anywhere, anytime, untethered access to the global Internet, is a similar explosion in the use of wireless Internet devices just around the corner?

Regardless of the future growth of wireless Internet devices, it's already clear that wireless networks and the mobility-related services they enable are here to stay. From a networking standpoint, the challenges posed by these networks, particularly at the link layer and the network layer, are so different from traditional wired computer networks that an individual chapter devoted to the study of wireless and mobile networks (i.e., *this* chapter) is appropriate.

We'll begin this chapter with a discussion of mobile users, wireless links, and networks, and their relationship to the larger (typically wired) networks to which they connect. We'll draw a distinction between the challenges posed by the *wireless* nature of the communication links in such networks, and by the *mobility* that these wireless links enable. Making this important distinction—between wireless and

mobility—will allow us to better isolate, identify, and master the key concepts in each area. Note that there are indeed many networked environments in which the network nodes are wireless but not mobile (e.g., wireless home or office networks with stationary workstations and large displays), and that there are limited forms of mobility that do not require wireless links (e.g., a worker who uses a wired laptop at home, shuts down the laptop, drives to work, and attaches the laptop to the company's wired network). Of course, many of the most exciting networked environments are those in which users are both wireless *and* mobile—for example, a scenario in which a mobile user (say in the back seat of car) maintains a Voice-over-IP call and multiple ongoing TCP connections while racing down the autobahn at 160 kilometers per hour. It is here, at the intersection of wireless and mobility, that we'll find the most interesting technical challenges!

We'll begin by illustrating the setting in which we'll consider wireless communication and mobility—a network in which wireless (and possibly mobile) users are connected into the larger network infrastructure by a wireless link at the network's edge. We'll then consider the characteristics of this wireless link in Section 6.2. We include a brief introduction to code division multiple access (CDMA), a shared-medium access protocol that is often used in wireless networks, in Section 6.2. In Section 6.3, we'll examine the link-level aspects of the IEEE 802.11 (WiFi) wireless LAN standard in some depth; we'll also say a few words about Bluetooth and other wireless personal area networks. In Section 6.4, we'll provide an overview of cellular Internet access, including 3G and emerging 4G cellular technologies that provide both voice and high-speed Internet access. In Section 6.5, we'll turn our attention to mobility, focusing on the problems of locating a mobile user, routing to the mobile user, and "handing off" the mobile user who dynamically moves from one point of attachment to the network to another. We'll examine how these mobility services are implemented in the mobile IP standard and in GSM, in Sections 6.6 and 6.7, respectively. Finally, we'll consider the impact of wireless links and mobility on transport-layer protocols and networked applications in Section 6.8.

6.1 Introduction

Figure 6.1 shows the setting in which we'll consider the topics of wireless data communication and mobility. We'll begin by keeping our discussion general enough to cover a wide range of networks, including both wireless LANs such as IEEE 802.11 and cellular networks such as a 3G network; we'll drill down into a more detailed discussion of specific wireless architectures in later sections. We can identify the following elements in a wireless network:

- *Wireless hosts.* As in the case of wired networks, hosts are the end-system devices that run applications. A **wireless host** might be a laptop, palmtop, smartphone, or desktop computer. The hosts themselves may or may not be mobile.

CASE HISTORY

PUBLIC WIFI ACCESS: COMING SOON TO A LAMP POST NEAR YOU?

WiFi hotspots—public locations where users can find 802.11 wireless access—are becoming increasingly common in hotels, airports, and cafés around the world. Most college campuses offer ubiquitous wireless access, and it's hard to find a hotel that doesn't offer wireless Internet access.

Over the past decade a number of cities have designed, deployed, and operated municipal WiFi networks. The vision of providing ubiquitous WiFi access to the community as a public service (much like streetlights)—helping to bridge the digital divide by providing Internet access to all citizens and to promote economic development—is compelling. Many cities around the world, including Philadelphia, Toronto, Hong Kong, Minneapolis, London, and Auckland, have plans to provide ubiquitous wireless within the city, or have already done so to varying degrees. The goal in Philadelphia was to "turn Philadelphia into the nation's largest WiFi hotspot and help to improve education, bridge the digital divide, enhance neighborhood development, and reduce the costs of government." The ambitious program—an agreement between the city, Wireless Philadelphia (a nonprofit entity), and the Internet Service Provider Earthlink—built an operational network of 802.11b hotspots on streetlamp pole arms and traffic control devices that covered 80 percent of the city. But financial and operational concerns caused the network to be sold to a group of private investors in 2008, who later sold the network back to the city in 2010. Other cities, such as Minneapolis, Toronto, Hong Kong, and Auckland, have had success with smaller-scale efforts.

The fact that 802.11 networks operate in the unlicensed spectrum (and hence can be deployed without purchasing expensive spectrum use rights) would seem to make them financially attractive. However, 802.11 access points (see Section 6.3) have much shorter ranges than 3G cellular base stations (see Section 6.4), requiring a larger number of deployed endpoints to cover the same geographic region. Cellular data networks providing Internet access, on the other hand, operate in the licensed spectrum. Cellular providers pay billions of dollars for spectrum access rights for their networks, making cellular data networks a business rather than municipal undertaking.

- *Wireless links.* A host connects to a base station (defined below) or to another wireless host through a **wireless communication link**. Different wireless link technologies have different transmission rates and can transmit over different distances. Figure 6.2 shows two key characteristics (coverage area and link rate) of the more popular wireless network standards. (The figure is only meant to provide a rough idea of these characteristics. For example, some of these types of networks are only now being deployed, and some link rates can increase or decrease beyond the values shown depending on distance, channel conditions, and the number of users in the wireless network.) We'll cover these standards

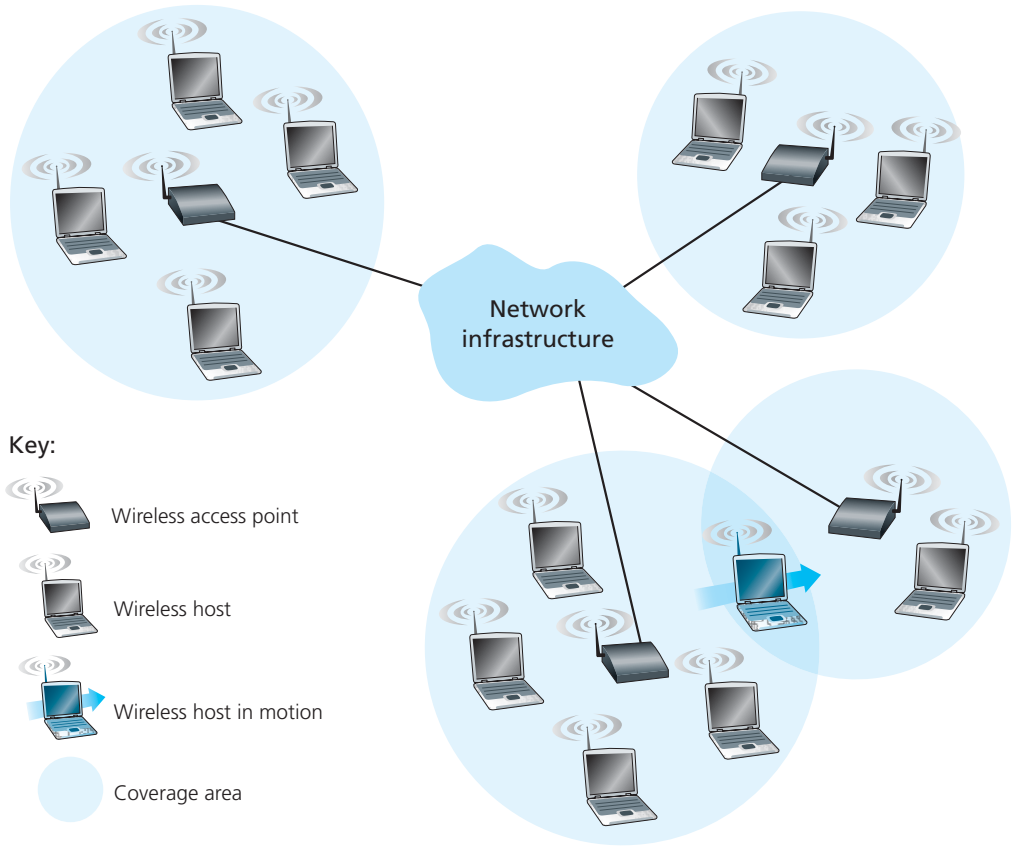


Figure 6.1 ♦ Elements of a wireless network

later in the first half of this chapter; we'll also consider other wireless link characteristics (such as their bit error rates and the causes of bit errors) in Section 6.2.

In Figure 6.1, wireless links connect wireless hosts located at the edge of the network into the larger network infrastructure. We hasten to add that wireless links are also sometimes used *within* a network to connect routers, switches, and other network equipment. However, our focus in this chapter will be on the use of wireless communication at the network edge, as it is here that many of the most exciting technical challenges, and most of the growth, are occurring.

- *Base station.* The **base station** is a key part of the wireless network infrastructure. Unlike the wireless host and wireless link, a base station has no obvious counterpart in a wired network. A base station is responsible for sending and receiving data (e.g., packets) to and from a wireless host that is associated with that base station. A base station will often be responsible for coordinating the transmission of multiple wireless hosts with which it is associated. When we say a wireless host is "associated" with a base station, we mean that (1) the host is within the wireless communication

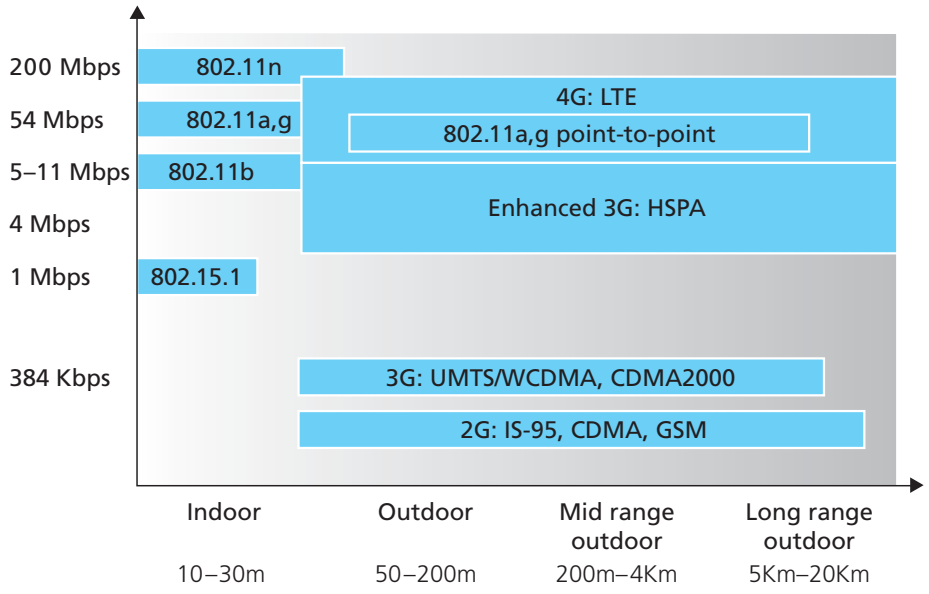


Figure 6.2 ♦ Link characteristics of selected wireless network standards

distance of the base station, and (2) the host uses that base station to relay data between it (the host) and the larger network. **Cell towers** in cellular networks and **access points** in 802.11 wireless LANs are examples of base stations.

In Figure 6.1, the base station is connected to the larger network (e.g., the Internet, corporate or home network, or telephone network), thus functioning as a link-layer relay between the wireless host and the rest of the world with which the host communicates.

Hosts associated with a base station are often referred to as operating in **infrastructure mode**, since all traditional network services (e.g., address assignment and routing) are provided by the network to which a host is connected via the base station. In **ad hoc networks**, wireless hosts have no such infrastructure with which to connect. In the absence of such infrastructure, the hosts themselves must provide for services such as routing, address assignment, DNS-like name translation, and more.

When a mobile host moves beyond the range of one base station and into the range of another, it will change its point of attachment into the larger network (i.e., change the base station with which it is associated)—a process referred to as **handoff**. Such mobility raises many challenging questions. If a host can move, how does one find the mobile host's current location in the network so that data can be forwarded to that mobile host? How is addressing performed, given that a host can be in one of many possible locations? If the host moves *during* a TCP

connection or phone call, how is data routed so that the connection continues uninterrupted? These and many (many!) other questions make wireless and mobile networking an area of exciting networking research.

- *Network infrastructure.* This is the larger network with which a wireless host may wish to communicate.

Having discussed the "pieces" of a wireless network, we note that these pieces can be combined in many different ways to form different types of wireless networks. You may find a taxonomy of these types of wireless networks useful as you read on in this chapter, or read/learn more about wireless networks beyond this book. At the highest level we can classify wireless networks according to two criteria: *(i)* whether a packet in the wireless network crosses exactly *one wireless hop or multiple wireless hops*, and *(ii)* whether there is *infrastructure* such as a base station in the network:

- *Single-hop, infrastructure-based.* These networks have a base station that is connected to a larger wired network (e.g., the Internet). Furthermore, all communication is between this base station and a wireless host over a single wireless hop. The 802.11 networks you use in the classroom, café, or library; and the 3G cellular data networks that we will learn about shortly all fall in this category.
- *Single-hop, infrastructure-less.* In these networks, there is no base station that is connected to a wireless network. However, as we will see, one of the nodes in this single-hop network may coordinate the transmissions of the other nodes. Bluetooth networks (which we will study in Section 6.3.6) and 802.11 networks in ad hoc mode are single-hop, infrastructure-less networks.
- *Multi-hop, infrastructure-based.* In these networks, a base station is present that is wired to the larger network. However, some wireless nodes may have to relay their communication through other wireless nodes in order to communicate via the base station. Some wireless sensor networks and so-called **wireless mesh networks** fall in this category.
- *Multi-hop, infrastructure-less.* There is no base station in these networks, and nodes may have to relay messages among several other nodes in order to reach a destination. Nodes may also be mobile, with connectivity changing among nodes—a class of networks known as **mobile ad hoc networks (MANETs)**. If the mobile nodes are vehicles, the network is a **vehicular ad hoc network (VANET)**. As you might imagine, the development of protocols for such networks is challenging and is the subject of much ongoing research.

In this chapter, we'll mostly confine ourselves to single-hop networks, and then mostly to infrastructure-based networks.

Let's now dig deeper into the technical challenges that arise in wireless and mobile networks. We'll begin by first considering the individual wireless link, deferring our discussion of mobility until later in this chapter.

6.2 Wireless Links and Network Characteristics

Let's begin by considering a simple wired network, say a home network, with a wired Ethernet switch (see Section 5.4) interconnecting the hosts. If we replace the wired Ethernet with a wireless 802.11 network, a wireless network interface would replace the host's wired Ethernet interface, and an access point would replace the Ethernet switch, but virtually no changes would be needed at the network layer or above. This suggests that we focus our attention on the link layer when looking for important differences between wired and wireless networks. Indeed, we can find a number of important differences between a wired link and a wireless link:

- *Decreasing signal strength.* Electromagnetic radiation attenuates as it passes through matter (e.g., a radio signal passing through a wall). Even in free space, the signal will disperse, resulting in decreased signal strength (sometimes referred to as **path loss**) as the distance between sender and receiver increases.
- *Interference from other sources.* Radio sources transmitting in the same frequency band will interfere with each other. For example, 2.4 GHz wireless phones and 802.11b wireless LANs transmit in the same frequency band. Thus, the 802.11b wireless LAN user talking on a 2.4 GHz wireless phone can expect that neither the network nor the phone will perform particularly well. In addition to interference from transmitting sources, electromagnetic noise within the environment (e.g., a nearby motor, a microwave) can result in interference.
- *Multipath propagation.* **Multipath propagation** occurs when portions of the electromagnetic wave reflect off objects and the ground, taking paths of different lengths between a sender and receiver. This results in the blurring of the received signal at the receiver. Moving objects between the sender and receiver can cause multipath propagation to change over time.

For a detailed discussion of wireless channel characteristics, models, and measurements, see [Anderson 1995].

The discussion above suggests that bit errors will be more common in wireless links than in wired links. For this reason, it is perhaps not surprising that wireless link protocols (such as the 802.11 protocol we'll examine in the following section) employ not only powerful CRC error detection codes, but also link-level reliable-data-transfer protocols that retransmit corrupted frames.

Having considered the impairments that can occur on a wireless channel, let's next turn our attention to the host receiving the wireless signal. This host receives an electromagnetic signal that is a combination of a degraded form of the original signal transmitted by the sender (degraded due to the attenuation and multipath propagation effects that we discussed above, among others) and background noise in the environment. The **signal-to-noise ratio (SNR)** is a relative measure of the strength of the received signal (i.e., the information being transmitted) and this noise. The SNR is typically measured in units of decibels (dB), a unit of measure that some think is used by electrical engineers primarily to confuse computer scientists. The SNR, measured in dB, is twenty times the ratio of the base-10 logarithm of the amplitude of the received signal to the amplitude of the noise. For our purposes here, we need only know that a larger SNR makes it easier for the receiver to extract the transmitted signal from the background noise.

Figure 6.3 (adapted from [Holland 2001]) shows the bit error rate (BER)—roughly speaking, the probability that a transmitted bit is received in error at the receiver—versus the SNR for three different modulation techniques for encoding information for transmission on an idealized wireless channel. The theory of modulation and coding, as well as signal extraction and BER, is well beyond the scope of this text (see [Schwartz 1980] for a discussion of these topics). Nonetheless, Figure 6.3 illustrates several physical-layer characteristics that are important in understanding higher-layer wireless communication protocols:

- *For a given modulation scheme, the higher the SNR, the lower the BER.* Since a sender can increase the SNR by increasing its transmission power, a sender

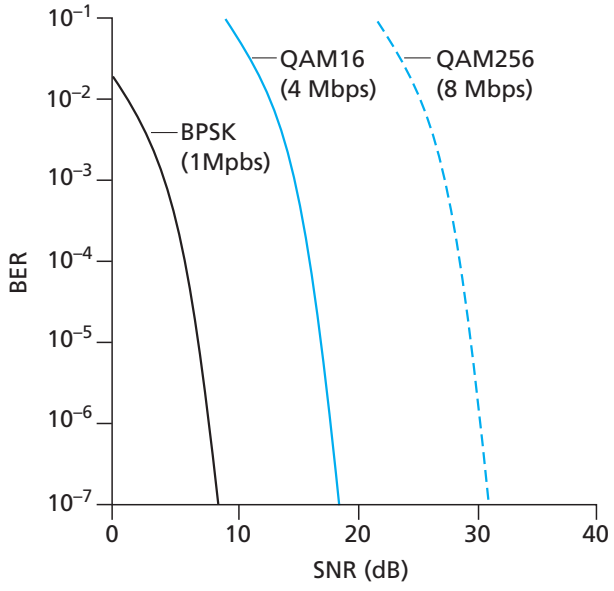


Figure 6.3 ♦ Bit error rate, transmission rate, and SNR

can decrease the probability that a frame is received in error by increasing its transmission power. Note, however, that there is arguably little practical gain in increasing the power beyond a certain threshold, say to decrease the BER from 10^{-12} to 10^{-13}. There are also *disadvantages* associated with increasing the transmission power: More energy must be expended by the sender (an important concern for battery-powered mobile users), and the sender's transmissions are more likely to interfere with the transmissions of another sender (see Figure 6.4(b)).

- *For a given SNR, a modulation technique with a higher bit transmission rate (whether in error or not) will have a higher BER.* For example, in Figure 6.3, with an SNR of 10 dB, BPSK modulation with a transmission rate of 1 Mbps has a BER of less than 10^{-7}, while with QAM16 modulation with a transmission rate of 4 Mbps, the BER is 10^{-1}, far too high to be practically useful. However, with an SNR of 20 dB, QAM16 modulation has a transmission rate of 4 Mbps and a BER of 10^{-7}, while BPSK modulation has a transmission rate of only 1 Mbps and a BER that is so low as to be (literally) "off the charts." If one can tolerate a BER of 10^{-7}, the higher transmission rate offered by QAM16 would make it the preferred modulation technique in this situation. These considerations give rise to the final characteristic, described next.
- *Dynamic selection of the physical-layer modulation technique can be used to adapt the modulation technique to channel conditions.* The SNR (and hence the BER) may change as a result of mobility or due to changes in the environment. Adaptive modulation and coding are used in cellular data systems and in the 802.11 WiFi and 3G cellular data networks that we'll study in Sections 6.3 and 6.4. This allows, for example, the selection of a modulation technique that provides the highest transmission rate possible subject to a constraint on the BER, for given channel characteristics.

A higher and time-varying bit error rate is not the only difference between a wired and wireless link. Recall that in the case of wired broadcast links, all nodes receive the transmissions from all other nodes. In the case of wireless links, the situation is not as simple, as shown in Figure 6.4. Suppose that Station A is transmitting to Station B. Suppose also that Station C is transmitting to Station B. With the so-called **hidden terminal problem**, physical obstructions in the environment (for example, a mountain or a building) may prevent A and C from hearing each other's transmissions, even though A's and C's transmissions are indeed interfering at the destination, B. This is shown in Figure 6.4(a). A second scenario that results in undetectable collisions at the receiver results from the **fading** of a signal's strength as it propagates through the wireless medium. Figure 6.4(b) illustrates the case where A and C are placed such that their signals are not strong enough to detect each other's transmissions, yet their signals *are* strong enough to interfere with each other at station B. As we'll see in Section 6.3, the hidden terminal problem and fading

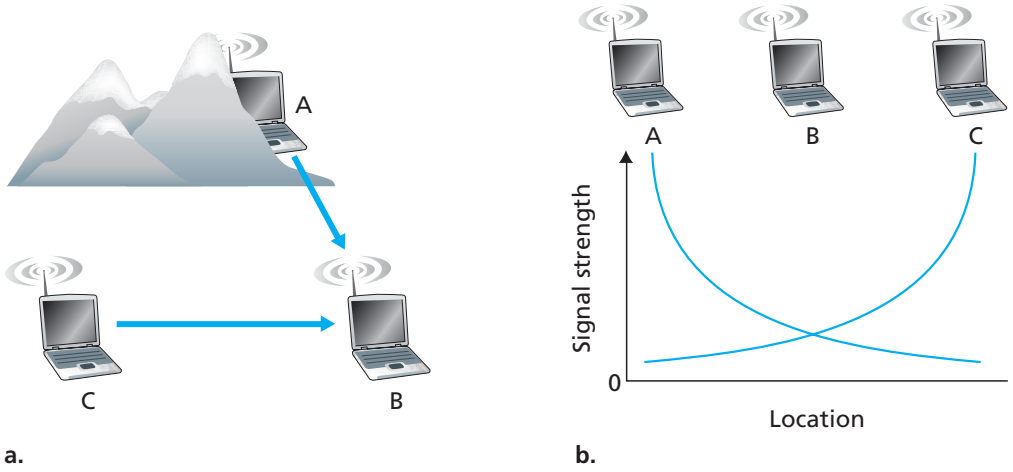


Figure 6.4 ♦ Hidden terminal problem caused by obstacle (a) and fading (b)

make multiple access in a wireless network considerably more complex than in a wired network.

6.2.1 CDMA

Recall from Chapter 5 that when hosts communicate over a shared medium, a protocol is needed so that the signals sent by multiple senders do not interfere at the receivers. In Chapter 5 we described three classes of medium access protocols: channel partitioning, random access, and taking turns. Code division multiple access (CDMA) belongs to the family of channel partitioning protocols. It is prevalent in wireless LAN and cellular technologies. Because CDMA is so important in the wireless world, we'll take a quick look at CDMA now, before getting into specific wireless access technologies in the subsequent sections.

In a CDMA protocol, each bit being sent is encoded by multiplying the bit by a signal (the code) that changes at a much faster rate (known as the **chipping rate**) than the original sequence of data bits. Figure 6.5 shows a simple, idealized CDMA encoding/decoding scenario. Suppose that the rate at which original data bits reach the CDMA encoder defines the unit of time; that is, each original data bit to be transmitted requires a one-bit slot time. Let d_i be the value of the data bit for the ith bit slot. For mathematical convenience, we represent a data bit with a 0 value as –1. Each bit slot is further subdivided into M mini-slots; in Figure 6.5, $M = 8$, although in practice M is much larger. The CDMA code used by the sender consists of a sequence of M values, c_m, $m = 1, \ldots, M$, each taking a +1 or –1

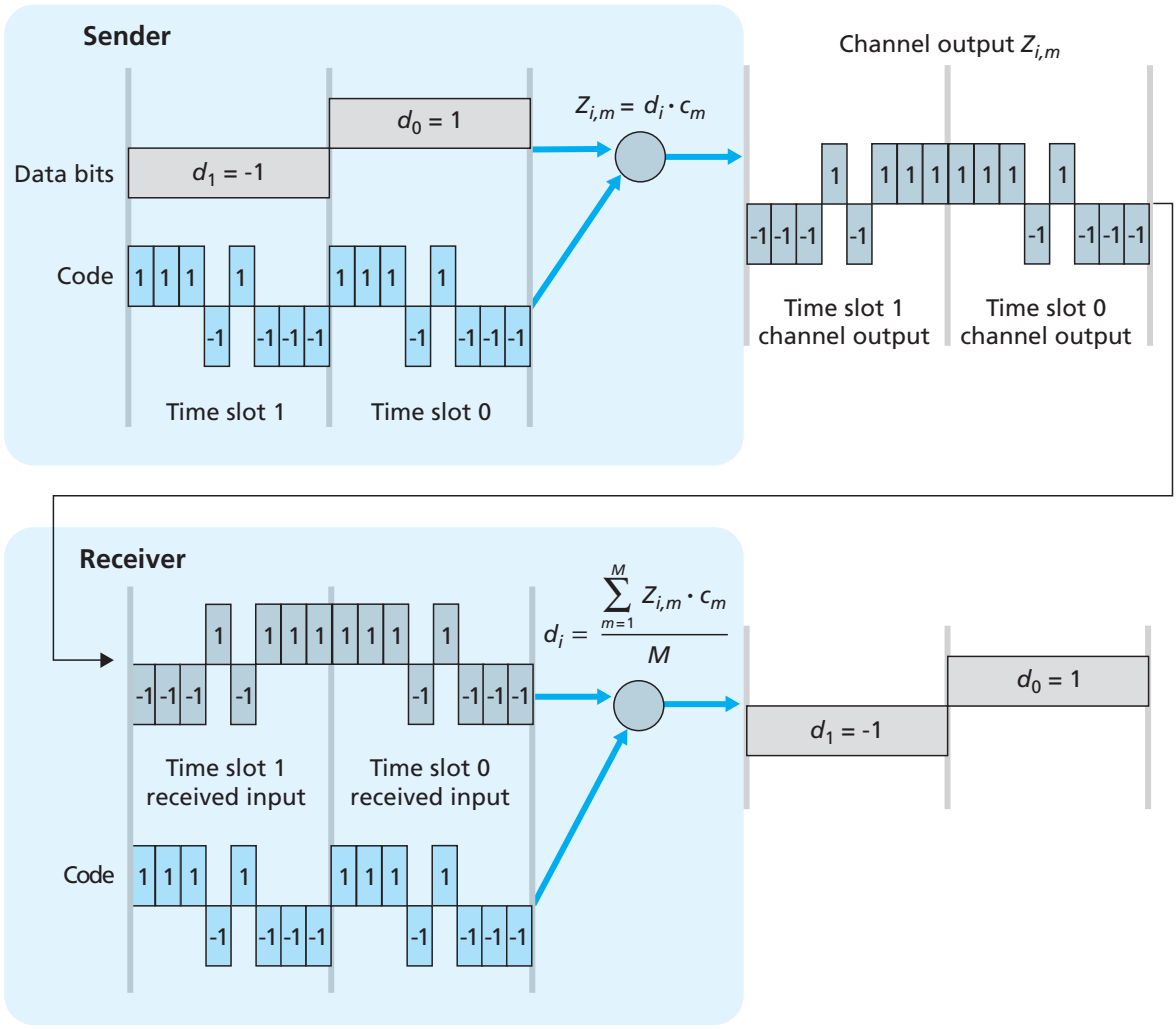


Figure 6.5 ♦ A simple CDMA example: sender encoding, receiver decoding

value. In the example in Figure 6.5, the M-bit CDMA code being used by the sender is (1, 1, 1, –1, 1, –1, –1, –1).

To illustrate how CDMA works, let us focus on the ith data bit, d_i. For the mth mini-slot of the bit-transmission time of d_i, the output of the CDMA encoder, $Z_{i,m}$, is the value of d_i multiplied by the mth bit in the assigned CDMA code, c_m:

$$Z_{i,m} = d_i \cdot c_m \tag{6.1}$$

In a simple world, with no interfering senders, the receiver would receive the encoded bits, $Z_{i,m}$, and recover the original data bit, d_i, by computing:

$$d_i = \frac{1}{M}\sum_{m=1}^{M} Z_{i,m} \cdot c_m \tag{6.2}$$

The reader might want to work through the details of the example in Figure 6.5 to see that the original data bits are indeed correctly recovered at the receiver using Equation 6.2.

The world is far from ideal, however, and as noted above, CDMA must work in the presence of interfering senders that are encoding and transmitting their data using a different assigned code. But how can a CDMA receiver recover a sender's original data bits when those data bits are being tangled with bits being transmitted by other senders? CDMA works under the assumption that the interfering transmitted bit signals are additive. This means, for example, that if three senders send a 1 value, and a fourth sender sends a –1 value during the same mini-slot, then the received signal at all receivers during that mini-slot is a 2 (since 1 + 1 + 1 − 1 = 2). In the presence of multiple senders, sender s computes its encoded transmissions, $Z^s_{i,m}$, in exactly the same manner as in Equation 6.1. The value received at a receiver during the mth mini-slot of the ith bit slot, however, is now the *sum* of the transmitted bits from all N senders during that mini-slot:

$$Z^*_{i,m} = \sum_{s=1}^{N} Z^s_{i,m}$$

Amazingly, if the senders' codes are chosen carefully, each receiver can recover the data sent by a given sender out of the aggregate signal simply by using the sender's code in exactly the same manner as in Equation 6.2:

$$d_i = \frac{1}{M}\sum_{m=1}^{M} Z^*_{i,m} \cdot c_m \tag{6.3}$$

as shown in Figure 6.6, for a two-sender CDMA example. The M-bit CDMA code being used by the upper sender is (1, 1, 1, –1, 1, –1, –1, –1), while the CDMA code being used by the lower sender is (1, –1, 1, 1, 1, –1, 1, 1). Figure 6.6 illustrates a receiver recovering the original data bits from the upper sender. Note that the receiver is able to extract the data from sender 1 in spite of the interfering transmission from sender 2.

Recall our cocktail analogy from Chapter 5. A CDMA protocol is similar to having partygoers speaking in multiple languages; in such circumstances humans are actually quite good at locking into the conversation in the language they understand, while filtering out the remaining conversations. We see here that CDMA is a partitioning protocol in that it partitions the codespace (as opposed to time or frequency) and assigns each node a dedicated piece of the codespace.

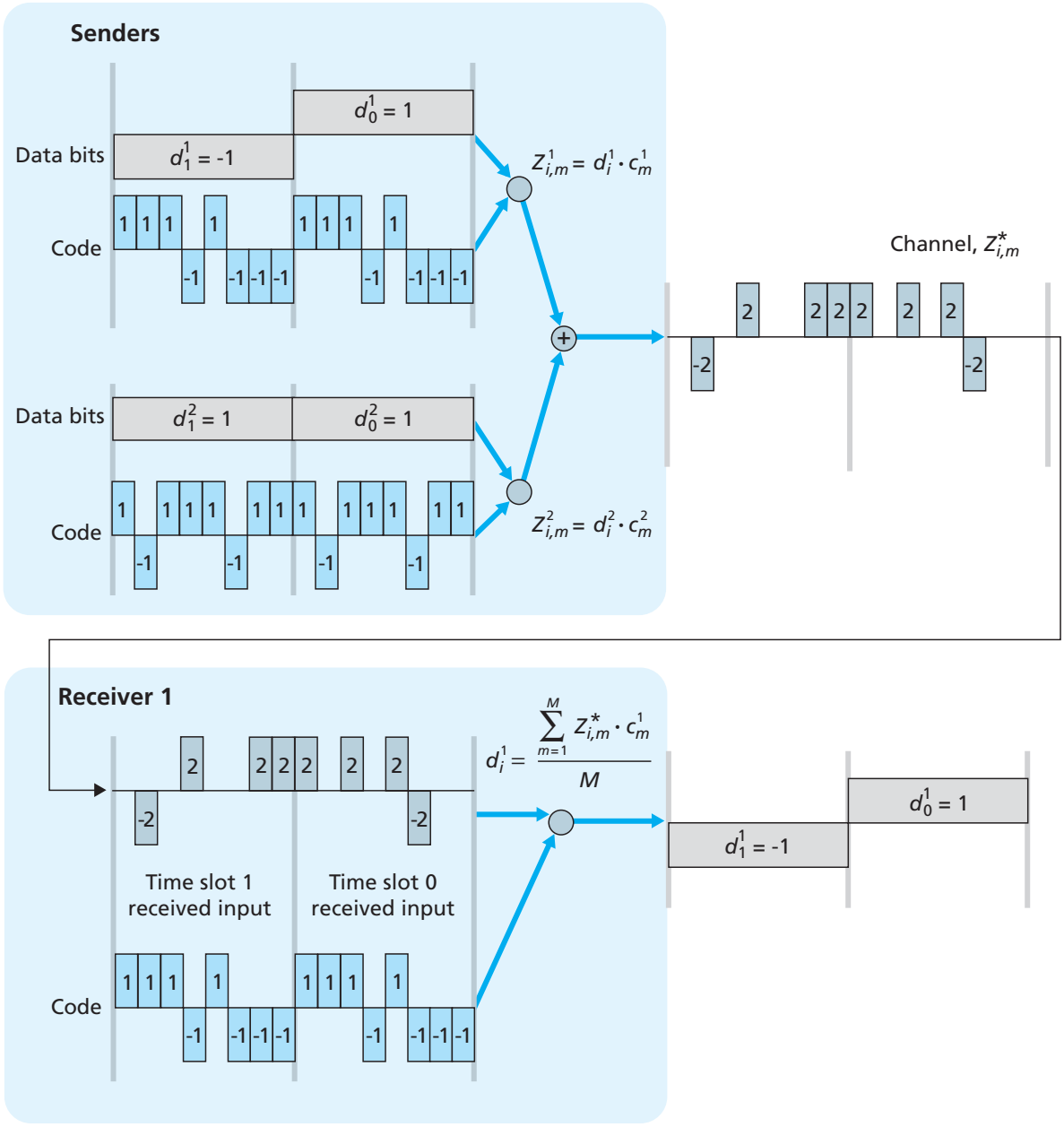


Figure 6.6 ♦ A two-sender CDMA example

Our discussion here of CDMA is necessarily brief; in practice a number of difficult issues must be addressed. First, in order for the CDMA receivers to be able to extract a particular sender's signal, the CDMA codes must be carefully chosen. Second, our discussion has assumed that the received signal strengths

from various senders are the same; in reality this can be difficult to achieve. There is a considerable body of literature addressing these and other issues related to CDMA; see [Pickholtz 1982; Viterbi 1995] for details.

6.3 WiFi: 802.11 Wireless LANs

Pervasive in the workplace, the home, educational institutions, cafés, airports, and street corners, wireless LANs are now one of the most important access network technologies in the Internet today. Although many technologies and standards for wireless LANs were developed in the 1990s, one particular class of standards has clearly emerged as the winner: the **IEEE 802.11 wireless LAN**, also known as **WiFi**. In this section, we'll take a close look at 802.11 wireless LANs, examining its frame structure, its medium access protocol, and its internetworking of 802.11 LANs with wired Ethernet LANs.

There are several 802.11 standards for wireless LAN technology, including 802.11b, 802.11a, and 802.11g. Table 6.1 summarizes the main characteristics of these standards. 802.11g is by far the most popular technology. A number of dual-mode (802.11a/g) and tri-mode (802.11a/b/g) devices are also available.

The three 802.11 standards share many characteristics. They all use the same medium access protocol, CSMA/CA, which we'll discuss shortly. All three use the same frame structure for their link-layer frames as well. All three standards have the ability to reduce their transmission rate in order to reach out over greater distances. And all three standards allow for both "infrastructure mode" and "ad hoc mode," as we'll also shortly discuss. However, as shown in Table 6.1, the three standards have some major differences at the physical layer.

The 802.11b wireless LAN has a data rate of 11 Mbps and operates in the unlicensed frequency band of 2.4–2.485 GHz, competing for frequency spectrum with 2.4 GHz phones and microwave ovens. 802.11a wireless LANs can run at

Standard	Frequency Range (United States)	Data Rate
802.11b	2.4–2.485 GHz	up to 11 Mbps
802.11a	5.1–5.8 GHz	up to 54 Mbps
802.11g	2.4–2.485 GHz	up to 54 Mbps

Table 6.1 ♦ Summary of IEEE 802.11 standards

significantly higher bit rates, but do so at higher frequencies. By operating at a higher frequency, 802.11a LANs have a shorter transmission distance for a given power level and suffer more from multipath propagation. 802.11g LANs, operating in the same lower-frequency band as 802.11b and being backwards compatible with 802.11b (so one can upgrade 802.11b clients incrementally) yet with the higher-speed transmission rates of 802.11a, allows users to have their cake and eat it too.

A relatively new WiFi standard, 802.11n [IEEE 802.11n 2012], uses multiple-input multiple-output (MIMO) antennas; i.e., two or more antennas on the sending side and two or more antennas on the receiving side that are transmitting/receiving different signals [Diggavi 2004]. Depending on the modulation scheme used, transmission rates of several hundred megabits per second are possible with 802.11n.

6.3.1 The 802.11 Architecture

Figure 6.7 illustrates the principal components of the 802.11 wireless LAN architecture. The fundamental building block of the 802.11 architecture is the **basic service set (BSS)**. A BSS contains one or more wireless stations and a central

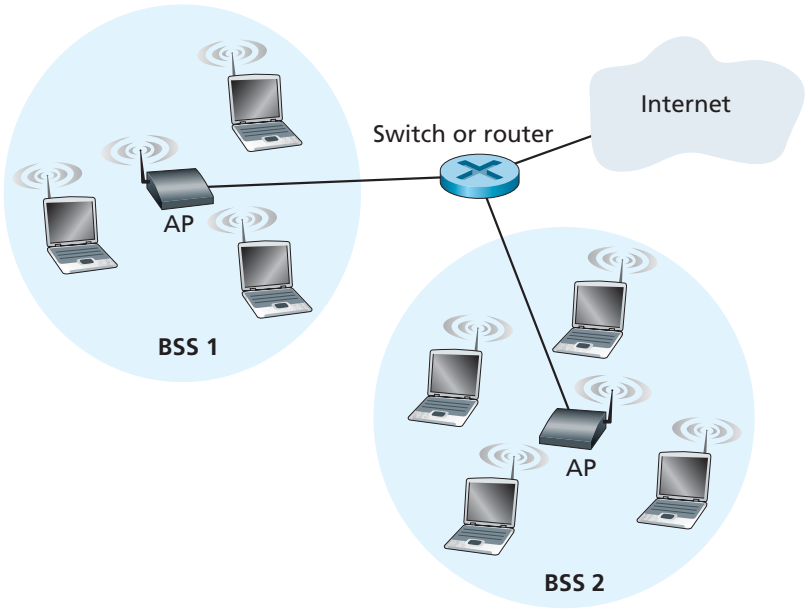


Figure 6.7 ♦ IEEE 802.11 LAN architecture

base station, known as an **access point (AP)** in 802.11 parlance. Figure 6.7 shows the AP in each of two BSSs connecting to an interconnection device (such as a switch or router), which in turn leads to the Internet. In a typical home network, there is one AP and one router (typically integrated together as one unit) that connects the BSS to the Internet.

As with Ethernet devices, each 802.11 wireless station has a 6-byte MAC address that is stored in the firmware of the station's adapter (that is, 802.11 network interface card). Each AP also has a MAC address for its wireless interface. As with Ethernet, these MAC addresses are administered by IEEE and are (in theory) globally unique.

As noted in Section 6.1, wireless LANs that deploy APs are often referred to as **infrastructure wireless LANs**, with the "infrastructure" being the APs along with the wired Ethernet infrastructure that interconnects the APs and a router. Figure 6.8 shows that IEEE 802.11 stations can also group themselves together to form an ad hoc network—a network with no central control and with no connections to the "outside world." Here, the network is formed "on the fly," by mobile devices that have found themselves in proximity to each other, that have a need to communicate, and that find no preexisting network infrastructure in their location. An ad hoc network might be formed when people with laptops get together (for example, in a conference room, a train, or a car) and want to exchange data in the absence of a centralized AP. There has been tremendous interest in ad hoc networking, as communicating portable devices continue to proliferate. In this section, though, we'll focus our attention on infrastructure wireless LANs.

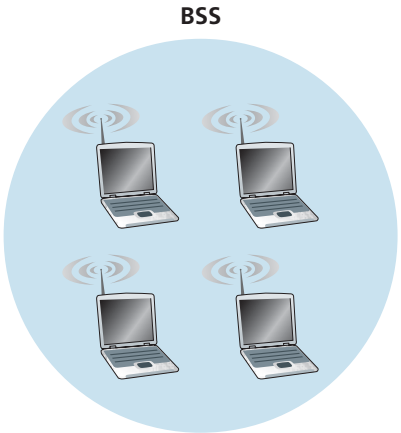


Figure 6.8 ♦ An IEEE 802.11 ad hoc network

Channels and Association

In 802.11, each wireless station needs to associate with an AP before it can send or receive network-layer data. Although all of the 802.11 standards use association, we'll discuss this topic specifically in the context of IEEE 802.11b/g.

When a network administrator installs an AP, the administrator assigns a one- or two-word **Service Set Identifier (SSID)** to the access point. (When you "view available networks" in Microsoft Windows XP, for example, a list is displayed showing the SSID of each AP in range.) The administrator must also assign a channel number to the AP. To understand channel numbers, recall that 802.11 operates in the frequency range of 2.4 GHz to 2.485 GHz. Within this 85 MHz band, 802.11 defines 11 partially overlapping channels. Any two channels are non-overlapping if and only if they are separated by four or more channels. In particular, the set of channels 1, 6, and 11 is the only set of three non-overlapping channels. This means that an administrator could create a wireless LAN with an aggregate maximum transmission rate of 33 Mbps by installing three 802.11b APs at the same physical location, assigning channels 1, 6, and 11 to the APs, and interconnecting each of the APs with a switch.

Now that we have a basic understanding of 802.11 channels, let's describe an interesting (and not completely uncommon) situation—that of a WiFi jungle. A **WiFi jungle** is any physical location where a wireless station receives a sufficiently strong signal from two or more APs. For example, in many cafés in New York City, a wireless station can pick up a signal from numerous nearby APs. One of the APs might be managed by the café, while the other APs might be in residential apartments near the café. Each of these APs would likely be located in a different IP subnet and would have been independently assigned a channel.

Now suppose you enter such a WiFi jungle with your portable computer, seeking wireless Internet access and a blueberry muffin. Suppose there are five APs in the WiFi jungle. To gain Internet access, your wireless station needs to join exactly one of the subnets and hence needs to **associate** with exactly one of the APs. Associating means the wireless station creates a virtual wire between itself and the AP. Specifically, only the associated AP will send data frames (that is, frames containing data, such as a datagram) to your wireless station, and your wireless station will send data frames into the Internet only through the associated AP. But how does your wireless station associate with a particular AP? And more fundamentally, how does your wireless station know which APs, if any, are out there in the jungle?

The 802.11 standard requires that an AP periodically send **beacon frames**, each of which includes the AP's SSID and MAC address. Your wireless station, knowing that APs are sending out beacon frames, scans the 11 channels, seeking beacon frames from any APs that may be out there (some of which may be transmitting on the same channel—it's a jungle out there!). Having learned about available APs

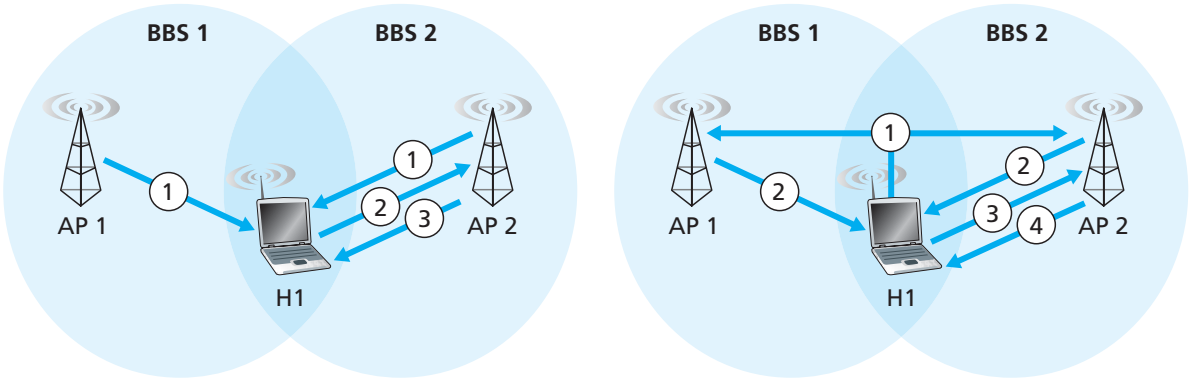
from the beacon frames, you (or your wireless host) select one of the APs for association.

The 802.11 standard does not specify an algorithm for selecting which of the available APs to associate with; that algorithm is left up to the designers of the 802.11 firmware and software in your wireless host. Typically, the host chooses the AP whose beacon frame is received with the highest signal strength. While a high signal strength is good (see, e.g., Figure 6.3), signal strength is not the only AP characteristic that will determine the performance a host receives. In particular, it's possible that the selected AP may have a strong signal, but may be overloaded with other affiliated hosts (that will need to share the wireless bandwidth at that AP), while an unloaded AP is not selected due to a slightly weaker signal. A number of alternative ways of choosing APs have thus recently been proposed [Vasudevan 2005; Nicholson 2006; Sundaresan 2006]. For an interesting and down-to-earth discussion of how signal strength is measured, see [Bardwell 2004].

The process of scanning channels and listening for beacon frames is known as **passive scanning** (see Figure 6.9a). A wireless host can also perform **active scanning**, by broadcasting a probe frame that will be received by all APs within the wireless host's range, as shown in Figure 6.9b. APs respond to the probe request frame with a probe response frame. The wireless host can then choose the AP with which to associate from among the responding APs.

After selecting the AP with which to associate, the wireless host sends an association request frame to the AP, and the AP responds with an association response frame. Note that this second request/response handshake is needed with active scanning, since an AP responding to the initial probe request frame doesn't know which of the (possibly many) responding APs the host will choose to associate with, in much the same way that a DHCP client can choose from among multiple DHCP servers (see Figure 4.21). Once associated with an AP, the host will want to join the subnet (in the IP addressing sense of Section 4.4.2) to which the AP belongs. Thus, the host will typically send a DHCP discovery message (see Figure 4.21) into the subnet via the AP in order to obtain an IP address on the subnet. Once the address is obtained, the rest of the world then views that host simply as another host with an IP address in that subnet.

In order to create an association with a particular AP, the wireless station may be required to authenticate itself to the AP. 802.11 wireless LANs provide a number of alternatives for authentication and access. One approach, used by many companies, is to permit access to a wireless network based on a station's MAC address. A second approach, used by many Internet cafés, employs usernames and passwords. In both cases, the AP typically communicates with an authentication server, relaying information between the wireless end-point station and the authentication server using a protocol such as RADIUS [RFC 2865] or DIAMETER [RFC 3588]. Separating the authentication server from the AP allows one authentication server to serve many APs, centralizing the (often sensitive) decisions of authentication and access within the single server, and keeping AP costs and complexity low. We'll see



a. Passive scanning
1. Beacon frames sent from APs
2. Association Request frame sent: H1 to selected AP
3. Association Response frame sent: Selected AP to H1

a. Active scanning
1. Probe Request frame broadcast from H1
2. Probes Response frame sent from APs
3. Association Request frame sent: H1 to selected AP
4. Association Response frame sent: Selected AP to H1

Figure 6.9 ♦ Active and passive scanning for access points

in Section 8.8 that the new IEEE 802.11i protocol defining security aspects of the 802.11 protocol family takes precisely this approach.

6.3.2 The 802.11 MAC Protocol

Once a wireless station is associated with an AP, it can start sending and receiving data frames to and from the access point. But because multiple stations may want to transmit data frames at the same time over the same channel, a multiple access protocol is needed to coordinate the transmissions. Here, a **station** is either a wireless station or an AP. As discussed in Chapter 5 and Section 6.2.1, broadly speaking there are three classes of multiple access protocols: channel partitioning (including CDMA), random access, and taking turns. Inspired by the huge success of Ethernet and its random access protocol, the designers of 802.11 chose a random access protocol for 802.11 wireless LANs. This random access protocol is referred to as **CSMA with collision avoidance**, or more succinctly as **CSMA/CA**. As with Ethernet's CSMA/CD, the "CSMA" in CSMA/CA stands for "carrier sense multiple access," meaning that each station senses the channel before transmitting, and refrains from transmitting when the channel is sensed busy. Although both Ethernet and 802.11 use carrier-sensing random access, the two MAC protocols have important differences.

First, instead of using collision detection, 802.11 uses collision-avoidance techniques. Second, because of the relatively high bit error rates of wireless channels, 802.11 (unlike Ethernet) uses a link-layer acknowledgment/retransmission (ARQ) scheme. We'll describe 802.11's collision-avoidance and link-layer acknowledgment schemes below.

Recall from Sections 5.3.2 and 5.4.2 that with Ethernet's collision-detection algorithm, an Ethernet station listens to the channel as it transmits. If, while transmitting, it detects that another station is also transmitting, it aborts its transmission and tries to transmit again after waiting a small, random amount of time. Unlike the 802.3 Ethernet protocol, the 802.11 MAC protocol does *not* implement collision detection. There are two important reasons for this:

- The ability to detect collisions requires the ability to send (the station's own signal) and receive (to determine whether another station is also transmitting) at the same time. Because the strength of the received signal is typically very small compared to the strength of the transmitted signal at the 802.11 adapter, it is costly to build hardware that can detect a collision.
- More importantly, even if the adapter could transmit and listen at the same time (and presumably abort transmission when it senses a busy channel), the adapter would still not be able to detect all collisions, due to the hidden terminal problem and fading, as discussed in Section 6.2.

Because 802.11wireless LANs do not use collision detection, once a station begins to transmit a frame, *it transmits the frame in its entirety*; that is, once a station gets started, there is no turning back. As one might expect, transmitting entire frames (particularly long frames) when collisions are prevalent can significantly degrade a multiple access protocol's performance. In order to reduce the likelihood of collisions, 802.11 employs several collision-avoidance techniques, which we'll shortly discuss.

Before considering collision avoidance, however, we'll first need to examine 802.11's **link-layer acknowledgment** scheme. Recall from Section 6.2 that when a station in a wireless LAN sends a frame, the frame may not reach the destination station intact for a variety of reasons. To deal with this non-negligible chance of failure, the 802.11 MAC protocol uses link-layer acknowledgments. As shown in Figure 6.10, when the destination station receives a frame that passes the CRC, it waits a short period of time known as the **Short Inter-frame Spacing (SIFS)** and then sends back an acknowledgment frame. If the transmitting station does not receive an acknowledgment within a given amount of time, it assumes that an error has occurred and retransmits the frame, using the CSMA/CA protocol to access the channel. If an acknowledgment is not received after some fixed number of retransmissions, the transmitting station gives up and discards the frame.

Having discussed how 802.11 uses link-layer acknowledgments, we're now in a position to describe the 802.11 CSMA/CA protocol. Suppose that a station (wireless station or an AP) has a frame to transmit.

1. If initially the station senses the channel idle, it transmits its frame after a short period of time known as the **Distributed Inter-frame Space (DIFS)**; see Figure 6.10.
2. Otherwise, the station chooses a random backoff value using binary exponential backoff (as we encountered in Section 5.3.2) and counts down this value when the channel is sensed idle. While the channel is sensed busy, the counter value remains frozen.
3. When the counter reaches zero (note that this can only occur while the channel is sensed idle), the station transmits the entire frame and then waits for an acknowledgment.

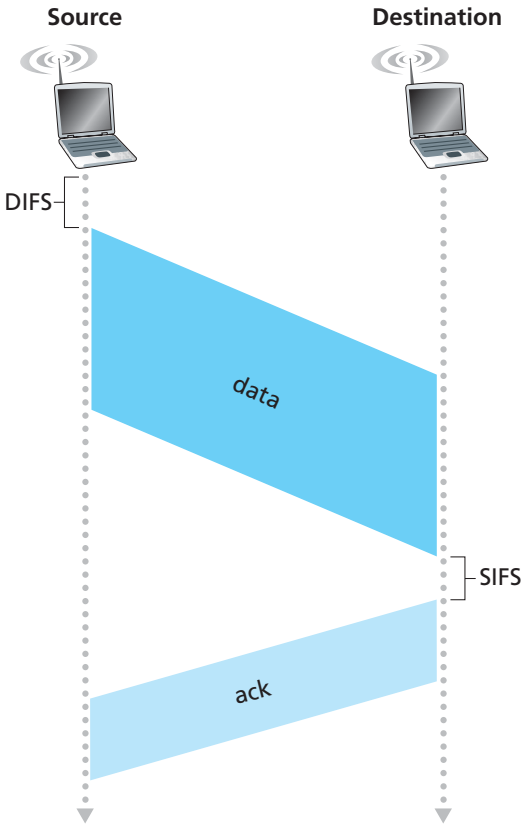


Figure 6.10 ♦ 802.11 uses link-layer acknowledgments

4. If an acknowledgment is received, the transmitting station knows that its frame has been correctly received at the destination station. If the station has another frame to send, it begins the CSMA/CA protocol at step 2. If the acknowledgment isn't received, the transmitting station reenters the backoff phase in step 2, with the random value chosen from a larger interval.

Recall that under Ethernet's CSMA/CD, multiple access protocol (Section 5.3.2), a station begins transmitting as soon as the channel is sensed idle. With CSMA/CA, however, the station refrains from transmitting while counting down, even when it senses the channel to be idle.Why do CSMA/CD and CDMA/CA take such different approaches here?

To answer this question, let's consider a scenario in which two stations each have a data frame to transmit, but neither station transmits immediately because each senses that a third station is already transmitting. With Ethernet's CSMA/CD, the two stations would each transmit as soon as they detect that the third station has finished transmitting. This would cause a collision, which isn't a serious issue in CSMA/CD, since both stations would abort their transmissions and thus avoid the useless transmissions of the remainders of their frames. In 802.11, however, the situation is quite different. Because 802.11 does not detect a collision and abort transmission, a frame suffering a collision will be transmitted in its entirety. The goal in 802.11 is thus to avoid collisions whenever possible. In 802.11, if the two stations sense the channel busy, they both immediately enter random backoff, hopefully choosing different backoff values. If these values are indeed different, once the channel becomes idle, one of the two stations will begin transmitting before the other, and (if the two stations are not hidden from each other) the "losing station" will hear the "winning station's" signal, freeze its counter, and refrain from transmitting until the winning station has completed its transmission. In this manner, a costly collision is avoided. Of course, collisions can still occur with 802.11 in this scenario: The two stations could be hidden from each other, or the two stations could choose random backoff values that are close enough that the transmission from the station starting first have yet to reach the second station. Recall that we encountered this problem earlier in our discussion of random access algorithms in the context of Figure 5.12.

Dealing with Hidden Terminals: RTS and CTS

The 802.11 MAC protocol also includes a nifty (but optional) reservation scheme that helps avoid collisions even in the presence of hidden terminals. Let's investigate this scheme in the context of Figure 6.11, which shows two wireless stations and one access point. Both of the wireless stations are within range of the AP (whose coverage is shown as a shaded circle) and both have associated with the AP. However, due to fading, the signal ranges of wireless stations are limited to the

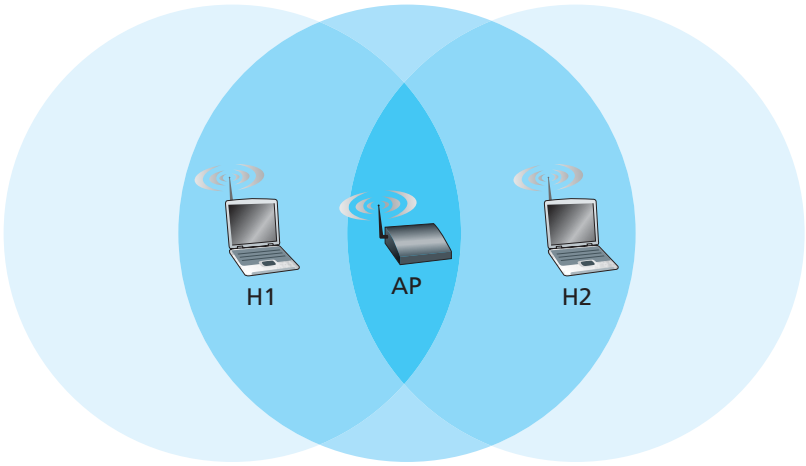


Figure 6.11 ♦ Hidden terminal example: H1 is hidden from H2, and vice versa

interiors of the shaded circles shown in Figure 6.11. Thus, each of the wireless stations is hidden from the other, although neither is hidden from the AP.

Let's now consider why hidden terminals can be problematic. Suppose Station H1 is transmitting a frame and halfway through H1's transmission, Station H2 wants to send a frame to the AP. H2, not hearing the transmission from H1, will first wait a DIFS interval and then transmit the frame, resulting in a collision. The channel will therefore be wasted during the entire period of H1's transmission as well as during H2's transmission.

In order to avoid this problem, the IEEE 802.11 protocol allows a station to use a short **Request to Send (RTS)** control frame and a short **Clear to Send (CTS)** control frame to *reserve* access to the channel. When a sender wants to send a DATA frame, it can first send an RTS frame to the AP, indicating the total time required to transmit the DATA frame and the acknowledgment (ACK) frame. When the AP receives the RTS frame, it responds by broadcasting a CTS frame. This CTS frame serves two purposes: It gives the sender explicit permission to send and also instructs the other stations not to send for the reserved duration.

Thus, in Figure 6.12, before transmitting a DATA frame, H1 first broadcasts an RTS frame, which is heard by all stations in its circle, including the AP. The AP then responds with a CTS frame, which is heard by all stations within its range, including H1 and H2. Station H2, having heard the CTS, refrains from transmitting for the time specified in the CTS frame. The RTS, CTS, DATA, and ACK frames are shown in Figure 6.12.

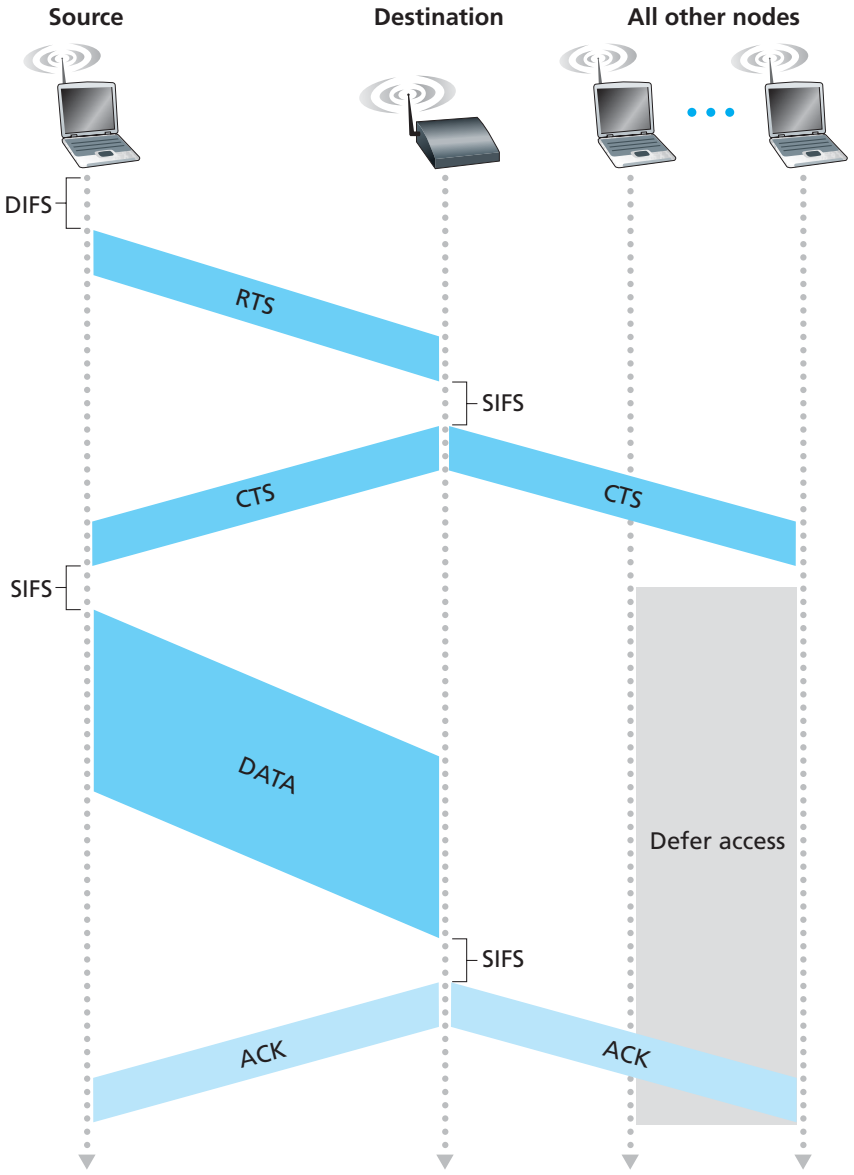


Figure 6.12 ♦ Collision avoidance using the RTS and CTS frames

The use of the RTS and CTS frames can improve performance in two important ways:

- The hidden station problem is mitigated, since a long DATA frame is transmitted only after the channel has been reserved.

- Because the RTS and CTS frames are short, a collision involving an RTS or CTS frame will last only for the duration of the short RTS or CTS frame. Once the RTS and CTS frames are correctly transmitted, the following DATA and ACK frames should be transmitted without collisions.

You are encouraged to check out the 802.11 applet in the textbook's companion Web site. This interactive applet illustrates the CSMA/CA protocol, including the RTS/CTS exchange sequence.

Although the RTS/CTS exchange can help reduce collisions, it also introduces delay and consumes channel resources. For this reason, the RTS/CTS exchange is only used (if at all) to reserve the channel for the transmission of a long DATA frame. In practice, each wireless station can set an RTS threshold such that the RTS/CTS sequence is used only when the frame is longer than the threshold. For many wireless stations, the default RTS threshold value is larger than the maximum frame length, so the RTS/CTS sequence is skipped for all DATA frames sent.

Using 802.11 as a Point-to-Point Link

Our discussion so far has focused on the use of 802.11 in a multiple access setting. We should mention that if two nodes each have a directional antenna, they can point their directional antennas at each other and run the 802.11 protocol over what is essentially a point-to-point link. Given the low cost of commodity 802.11 hardware, the use of directional antennas and an increased transmission power allow 802.11 to be used as an inexpensive means of providing wireless point-to-point connections over tens of kilometers distance. [Raman 2007] describes such a multi-hop wireless network operating in the rural Ganges plains in India that contains point-to-point 802.11 links.

6.3.3 The IEEE 802.11 Frame

Although the 802.11 frame shares many similarities with an Ethernet frame, it also contains a number of fields that are specific to its use for wireless links. The 802.11 frame is shown in Figure 6.13. The numbers above each of the fields in the frame represent the lengths of the fields in *bytes*; the numbers above each of the subfields in the frame control field represent the lengths of the subfields in *bits*. Let's now examine the fields in the frame as well as some of the more important subfields in the frame's control field.

Payload and CRC Fields

At the heart of the frame is the payload, which typically consists of an IP datagram or an ARP packet. Although the field is permitted to be as long as 2,312 bytes, it is

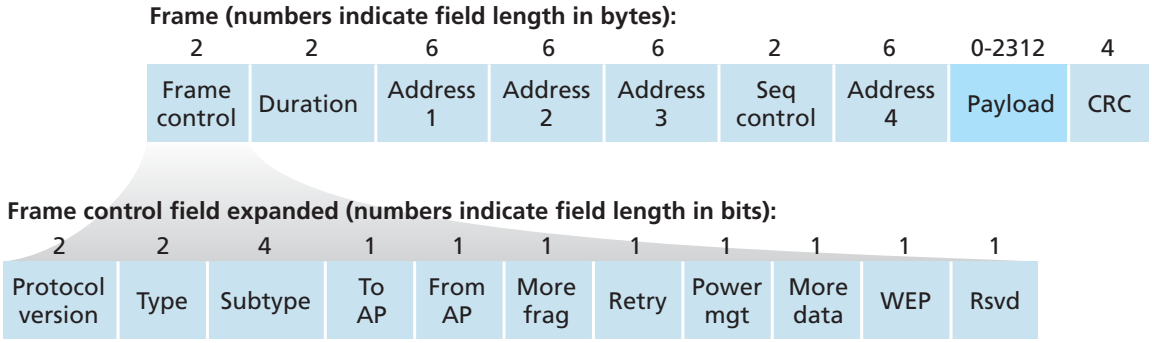


Figure 6.13 ♦ The 802.11 frame

typically fewer than 1,500 bytes, holding an IP datagram or an ARP packet. As with an Ethernet frame, an 802.11 frame includes a 32-bit cyclic redundancy check (CRC) so that the receiver can detect bit errors in the received frame. As we've seen, bit errors are much more common in wireless LANs than in wired LANs, so the CRC is even more useful here.

Address Fields

Perhaps the most striking difference in the 802.11 frame is that it has *four* address fields, each of which can hold a 6-byte MAC address. But why four address fields? Doesn't a source MAC field and destination MAC field suffice, as they do for Ethernet? It turns out that three address fields are needed for internetworking purposes—specifically, for moving the network-layer datagram from a wireless station through an AP to a router interface. The fourth address field is used when APs forward frames to each other in ad hoc mode. Since we are only considering infrastructure networks here, let's focus our attention on the first three address fields. The 802.11 standard defines these fields as follows:

- Address 2 is the MAC address of the station that transmits the frame. Thus, if a wireless station transmits the frame, that station's MAC address is inserted in the address 2 field. Similarly, if an AP transmits the frame, the AP's MAC address is inserted in the address 2 field.
- Address 1 is the MAC address of the wireless station that is to receive the frame. Thus if a mobile wireless station transmits the frame, address 1 contains the MAC address of the destination AP. Similarly, if an AP transmits the frame, address 1 contains the MAC address of the destination wireless station.

- To understand address 3, recall that the BSS (consisting of the AP and wireless stations) is part of a subnet, and that this subnet connects to other subnets via some router interface. Address 3 contains the MAC address of this router interface.

To gain further insight into the purpose of address 3, let's walk through an internetworking example in the context of Figure 6.14. In this figure, there are two APs, each of which is responsible for a number of wireless stations. Each of the APs has a direct connection to a router, which in turn connects to the global Internet. We should keep in mind that an AP is a link-layer device, and thus neither "speaks" IP nor understands IP addresses. Consider now moving a datagram from the router interface R1 to the wireless Station H1. The router is not aware that there is an AP between it and H1; from the router's perspective, H1 is just a host in one of the subnets to which it (the router) is connected.

- The router, which knows the IP address of H1 (from the destination address of the datagram), uses ARP to determine the MAC address of H1, just as in an ordinary Ethernet LAN. After obtaining H1's MAC address, router interface R1 encapsulates the datagram within an Ethernet frame. The source address field of this frame contains R1's MAC address, and the destination address field contains H1's MAC address.

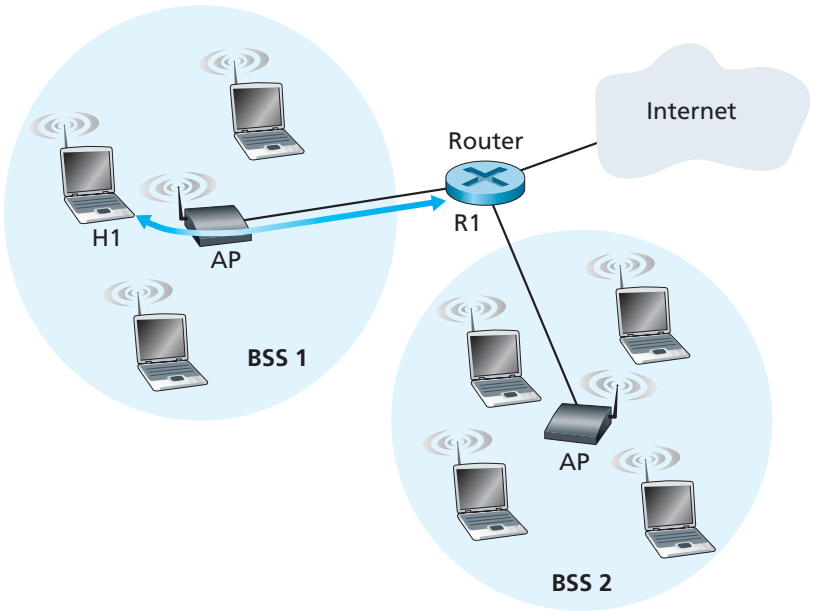


Figure 6.14 ♦ The use of address fields in 802.11 frames: Sending frames between H1 and R1

- When the Ethernet frame arrives at the AP, the AP converts the 802.3 Ethernet frame to an 802.11 frame before transmitting the frame into the wireless channel. The AP fills in address 1 and address 2 with H1's MAC address and its own MAC address, respectively, as described above. For address 3, the AP inserts the MAC address of R1. In this manner, H1 can determine (from address 3) the MAC address of the router interface that sent the datagram into the subnet.

Now consider what happens when the wireless station H1 responds by moving a datagram from H1 to R1.

- H1 creates an 802.11 frame, filling the fields for address 1 and address 2 with the AP's MAC address and H1's MAC address, respectively, as described above. For address 3, H1 inserts R1's MAC address.
- When the AP receives the 802.11 frame, it converts the frame to an Ethernet frame. The source address field for this frame is H1's MAC address, and the destination address field is R1's MAC address. Thus, address 3 allows the AP to determine the appropriate destination MAC address when constructing the Ethernet frame.

In summary, address 3 plays a crucial role for internetworking the BSS with a wired LAN.

Sequence Number, Duration, and Frame Control Fields

Recall that in 802.11, whenever a station correctly receives a frame from another station, it sends back an acknowledgment. Because acknowledgments can get lost, the sending station may send multiple copies of a given frame. As we saw in our discussion of the rdt2.1 protocol (Section 3.4.1), the use of sequence numbers allows the receiver to distinguish between a newly transmitted frame and the retransmission of a previous frame. The sequence number field in the 802.11 frame thus serves exactly the same purpose here at the link layer as it did in the transport layer in Chapter 3.

Recall that the 802.11 protocol allows a transmitting station to reserve the channel for a period of time that includes the time to transmit its data frame and the time to transmit an acknowledgment. This duration value is included in the frame's duration field (both for data frames and for the RTS and CTS frames).

As shown in Figure 6.13, the frame control field includes many subfields. We'll say just a few words about some of the more important subfields; for a more complete discussion, you are encouraged to consult the 802.11 specification [Held 2001; Crow 1997; IEEE 802.11 1999]. The *type* and *subtype* fields are used to distinguish the association, RTS, CTS, ACK, and data frames. The *to* and *from* fields are used to define the meanings of the different address fields. (These meanings change

depending on whether ad hoc or infrastructure modes are used and, in the case of infrastructure mode, whether a wireless station or an AP is sending the frame.) Finally the WEP field indicates whether encryption is being used or not. (WEP is discussed in Chapter 8.)

6.3.4 Mobility in the Same IP Subnet

In order to increase the physical range of a wireless LAN, companies and universities will often deploy multiple BSSs within the same IP subnet. This naturally raises the issue of mobility among the BSSs—how do wireless stations seamlessly move from one BSS to another while maintaining ongoing TCP sessions? As we'll see in this subsection, mobility can be handled in a relatively straightforward manner when the BSSs are part of the subnet. When stations move between subnets, more sophisticated mobility management protocols will be needed, such as those we'll study in Sections 6.5 and 6.6.

Let's now look at a specific example of mobility between BSSs in the same subnet. Figure 6.15 shows two interconnected BSSs with a host, H1, moving from BSS1 to BSS2. Because in this example the interconnection device that connects the two BSSs is *not* a router, all of the stations in the two BSSs, including the APs, belong to the same IP subnet. Thus, when H1 moves from BSS1 to BSS2, it may keep its IP address and all of its ongoing TCP connections. If the interconnection device were a router, then H1 would have to obtain a new IP address in the subnet in which it was moving. This address change would disrupt (and eventually terminate) any on-going TCP connections at H1. In Section 6.6, we'll see how a network-layer mobility protocol, such as mobile IP, can be used to avoid this problem.

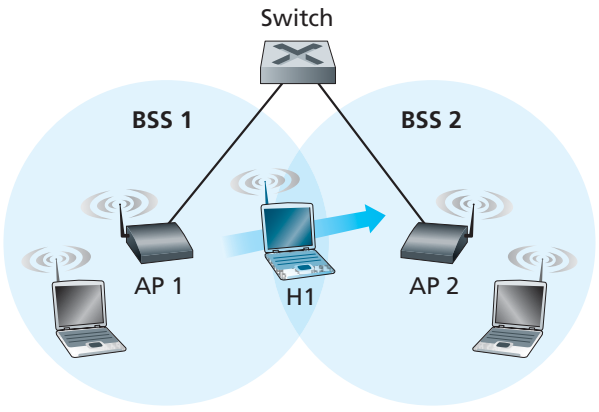


Figure 6.15 ♦ Mobility in the same subnet

But what specifically happens when H1 moves from BSS1 to BSS2? As H1 wanders away from AP1, H1 detects a weakening signal from AP1 and starts to scan for a stronger signal. H1 receives beacon frames from AP2 (which in many corporate and university settings will have the same SSID as AP1). H1 then disassociates with AP1 and associates with AP2, while keeping its IP address and maintaining its ongoing TCP sessions.

This addresses the handoff problem from the host and AP viewpoint. But what about the switch in Figure 6.15? How does it know that the host has moved from one AP to another? As you may recall from Chapter 5, switches are "self-learning" and automatically build their forwarding tables. This self-learning feature nicely handles occasional moves (for example, when an employee gets transferred from one department to another); however, switches were not designed to support highly mobile users who want to maintain TCP connections while moving between BSSs. To appreciate the problem here, recall that before the move, the switch has an entry in its forwarding table that pairs H1's MAC address with the outgoing switch interface through which H1 can be reached. If H1 is initially in BSS1, then a datagram destined to H1 will be directed to H1 via AP1. Once H1 associates with BSS2, however, its frames should be directed to AP2. One solution (a bit of a hack, really) is for AP2 to send a broadcast Ethernet frame with H1's source address to the switch just after the new association. When the switch receives the frame, it updates its forwarding table, allowing H1 to be reached via AP2. The 802.11f standards group is developing an inter-AP protocol to handle these and related issues.

6.3.5 Advanced Features in 802.11

We'll wrap up our coverage of 802.11 with a short discussion of two advanced capabilities found in 802.11 networks. As we'll see, these capabilities are *not* completely specified in the 802.11 standard, but rather are made possible by mechanisms specified in the standard. This allows different vendors to implement these capabilities using their own (proprietary) approaches, presumably giving them an edge over the competition.

802.11 Rate Adaptation

We saw earlier in Figure 6.3 that different modulation techniques (with the different transmission rates that they provide) are appropriate for different SNR scenarios. Consider for example a mobile 802.11 user who is initially 20 meters away from the base station, with a high signal-to-noise ratio. Given the high SNR, the user can communicate with the base station using a physical-layer modulation technique that provides high transmission rates while maintaining a low BER. This is one happy user! Suppose now that the user becomes mobile, walking away from

the base station, with the SNR falling as the distance from the base station increases. In this case, if the modulation technique used in the 802.11 protocol operating between the base station and the user does not change, the BER will become unacceptably high as the SNR decreases, and eventually no transmitted frames will be received correctly.

For this reason, some 802.11 implementations have a rate adaptation capability that adaptively selects the underlying physical-layer modulation technique to use based on current or recent channel characteristics. If a node sends two frames in a row without receiving an acknowledgment (an implicit indication of bit errors on the channel), the transmission rate falls back to the next lower rate. If 10 frames in a row are acknowledged, or if a timer that tracks the time since the last fallback expires, the transmission rate increases to the next higher rate. This rate adaptation mechanism shares the same "probing" philosophy as TCP's congestion-control mechanism—when conditions are good (reflected by ACK receipts), the transmission rate is increased until something "bad" happens (the lack of ACK receipts); when something "bad" happens, the transmission rate is reduced. 802.11 rate adaptation and TCP congestion control are thus similar to the young child who is constantly pushing his/her parents for more and more (say candy for a young child, later curfew hours for the teenager) until the parents finally say "Enough!" and the child backs off (only to try again later after conditions have hopefully improved!). A number of other schemes have also been proposed to improve on this basic automatic rate-adjustment scheme [Kamerman 1997; Holland 2001; Lacage 2004].

Power Management

Power is a precious resource in mobile devices, and thus the 802.11 standard provides power-management capabilities that allow 802.11 nodes to minimize the amount of time that their sense, transmit, and receive functions and other circuitry need to be "on." 802.11 power management operates as follows. A node is able to explicitly alternate between sleep and wake states (not unlike a sleepy student in a classroom!). A node indicates to the access point that it will be going to sleep by setting the power-management bit in the header of an 802.11 frame to 1. A timer in the node is then set to wake up the node just before the AP is scheduled to send its beacon frame (recall that an AP typically sends a beacon frame every 100 msec). Since the AP knows from the set power-transmission bit that the node is going to sleep, it (the AP) knows that it should not send any frames to that node, and will buffer any frames destined for the sleeping host for later transmission.

A node will wake up just before the AP sends a beacon frame, and quickly enter the fully active state (unlike the sleepy student, this wakeup requires only 250 microseconds [Kamerman 1997]!). The beacon frames sent out by the AP contain a list of nodes whose frames have been buffered at the AP. If there are no buffered frames for the node, it can go back to sleep. Otherwise, the node can explicitly

request that the buffered frames be sent by sending a polling message to the AP. With an inter-beacon time of 100 msec, a wakeup time of 250 microseconds, and a similarly small time to receive a beacon frame and check to ensure that there are no buffered frames, a node that has no frames to send or receive can be asleep 99% of the time, resulting in a significant energy savings.

6.3.6 Personal Area Networks: Bluetooth and Zigbee

As illustrated in Figure 6.2, the IEEE 802.11 WiFi standard is aimed at communication among devices separated by up to 100 meters (except when 802.11 is used in a point-to-point configuration with a directional antenna). Two other IEEE 802 protocols—Bluetooth and Zigbee (defined in the IEEE 802.15.1 and IEEE 802.15.4 standards [IEEE 802.15 2012]) and WiMAX (defined in the IEEE 802.16 standard [IEEE 802.16d 2004; IEEE 802.16e 2005])—are standards for communicating over shorter and longer distances, respectively. We will touch on WiMAX briefly when we discuss cellular data networks in Section 6.4, and so here, we will focus on networks for shorter distances.

Bluetooth

An IEEE 802.15.1 network operates over a short range, at low power, and at low cost. It is essentially a low-power, short-range, low-rate "cable replacement" technology for interconnecting notebooks, peripheral devices, cellular phones, and smartphones, whereas 802.11 is a higher-power, medium-range, higher-rate "access" technology. For this reason, 802.15.1 networks are sometimes referred to as wireless personal area networks (WPANs). The link and physical layers of 802.15.1 are based on the earlier **Bluetooth** specification for personal area networks [Held 2001, Bisdikian 2001]. 802.15.1 networks operate in the 2.4 GHz unlicensed radio band in a TDM manner, with time slots of 625 microseconds. During each time slot, a sender transmits on one of 79 channels, with the channel changing in a known but pseudo-random manner from slot to slot. This form of channel hopping, known as **frequency-hopping spread spectrum (FHSS),** spreads transmissions in time over the frequency spectrum. 802.15.1 can provide data rates up to 4 Mbps.

802.15.1 networks are ad hoc networks: No network infrastructure (e.g., an access point) is needed to interconnect 802.15.1 devices. Thus, 802.15.1 devices must organize themselves. 802.15.1 devices are first organized into a **piconet** of up to eight active devices, as shown in Figure 6.16. One of these devices is designated as the master, with the remaining devices acting as slaves. The master node truly rules the piconet—its clock determines time in the piconet, it can transmit in each odd-numbered slot, and a slave can transmit only after the master has communicated with it in the previous slot and even then the slave can only transmit to the master. In addition to the slave devices, there can also be up to 255 parked devices in the

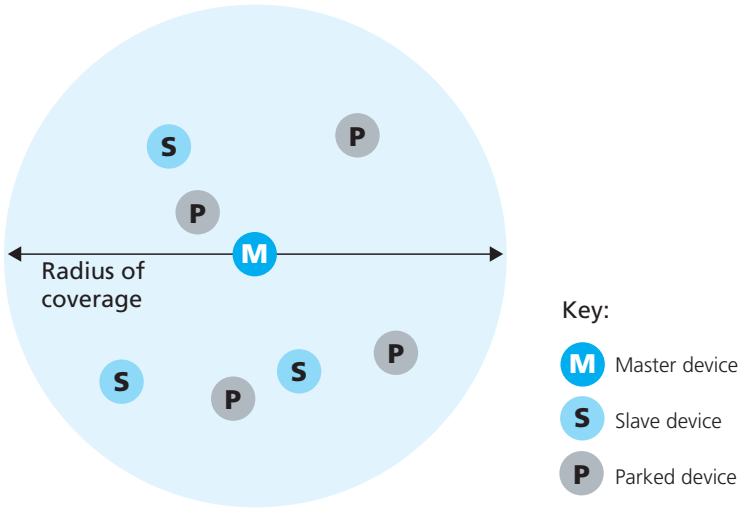


Figure 6.16 ♦ A Bluetooth piconet

network. These devices cannot communicate until their status has been changed from parked to active by the master node.

For more information about 802.15.1 WPANs, the interested reader should consult the Bluetooth references [Held 2001, Bisdikian 2001] or the official IEEE 802.15 Web site [IEEE 802.15 2012].

Zigbee

A second personal area network standardized by the IEEE is the 802.14.5 standard [IEEE 802.15 2012] known as Zigbee. While Bluetooth networks provide a "cable replacement" data rate of over a Megabit per second, Zigbee is targeted at lower-powered, lower-data-rate, lower-duty-cycle applications than Bluetooth. While we may tend to think that "bigger and faster is better," not all network applications need high bandwidth and the consequent higher costs (both economic and power costs). For example, home temperature and light sensors, security devices, and wall-mounted switches are all very simple, low-power, low-duty-cycle, low-cost devices. Zigbee is thus well-suited for these devices. Zigbee defines channel rates of 20, 40, 100, and 250 Kbps, depending on the channel frequency.

Nodes in a Zigbee network come in two flavors. So-called "reduced-function devices" operate as slave devices under the control of a single "full-function device," much as Bluetooth slave devices. A full-function device can operate as a master device as in Bluetooth by controlling multiple slave devices, and multiple full-function devices can additionally be configured into a mesh network in which full-function devices route frames amongst themselves. Zigbee shares

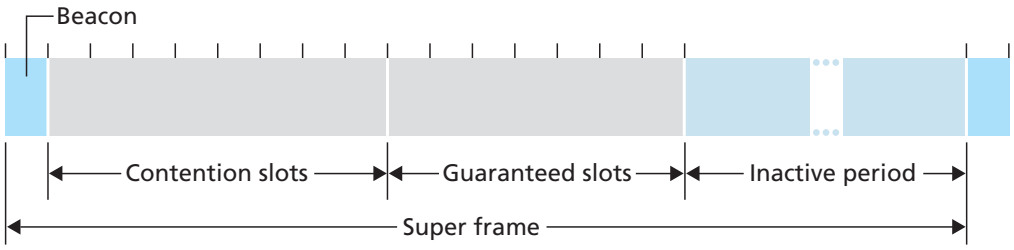


Figure 6.17 ♦ Zigbee 802.14.4 super-frame structure

many protocol mechanisms that we've already encountered in other link-layer protocols: beacon frames and link-layer acknowledgments (similar to 802.11), carrier-sense random access protocols with binary exponential backoff (similar to 802.11 and Ethernet), and fixed, guaranteed allocation of time slots (similar to DOCSIS).

Zigbee networks can be configured in many different ways. Let's consider the simple case of a single full-function device controlling multiple reduced-function devices in a time-slotted manner using beacon frames. Figure 6.17 shows the case where the Zigbee network divides time into recurring super frames, each of which begins with a beacon frame. Each beacon frame divides the super frame into an active period (during which devices may transmit) and an inactive period (during which all devices, including the controller, can sleep and thus conserve power). The active period consists of 16 time slots, some of which are used by devices in a CSMA/CA random access manner, and some of which are allocated by the controller to specific devices, thus providing guaranteed channel access for those devices. More details about Zigbee networks can be found at [Baronti 2007, IEEE 802.15.4 2012].

6.4 Cellular Internet Access

In the previous section we examined how an Internet host can access the Internet when inside a WiFi hotspot—that is, when it is within the vicinity of an 802.11 access point. But most WiFi hotspots have a small coverage area of between 10 and 100 meters in diameter. What do we do then when we have a desperate need for wireless Internet access and we cannot access a WiFi hotspot?

Given that cellular telephony is now ubiquitous in many areas throughout the world, a natural strategy is to extend cellular networks so that they support not only voice telephony but wireless Internet access as well. Ideally, this Internet access would be at a reasonably high speed and would provide for seamless

mobility, allowing users to maintain their TCP sessions while traveling, for example, on a bus or a train. With sufficiently high upstream and downstream bit rates, the user could even maintain video-conferencing sessions while roaming about. This scenario is not that far-fetched. As of 2012, many cellular telephony providers in the U.S. offer their subscribers a cellular Internet access service for under $50 per month with typical downstream and upstream bit rates in the hundreds of kilobits per second. Data rates of several megabits per second are becoming available as broadband data services such as those we will cover here become more widely deployed.

In this section, we provide a brief overview of current and emerging cellular Internet access technologies. Our focus here will be on both the wireless first hop as well as the network that connects the wireless first hop into the larger telephone network and/or the Internet; in Section 6.7 we'll consider how calls are routed to a user moving between base stations. Our brief discussion will necessarily provide only a simplified and high-level description of cellular technologies. Modern cellular communications, of course, has great breadth and depth, with many universities offering several courses on the topic. Readers seeking a deeper understanding are encouraged to see [Goodman 1997; Kaaranen 2001; Lin 2001; Korhonen 2003; Schiller 2003; Scourias 2012; Turner 2012; Akyildiz 2010], as well as the particularly excellent and exhaustive reference [Mouly 1992].

6.4.1 An Overview of Cellular Network Architecture

In our description of cellular network architecture in this section, we'll adopt the terminology of the *Global System for Mobile Communications (**GSM**)* standards. (For history buffs, the GSM acronym was originally derived from *Groupe Spécial Mobile*, until the more anglicized name was adopted, preserving the original acronym letters.) In the 1980s, Europeans recognized the need for a pan-European digital cellular telephony system that would replace the numerous incompatible analog cellular telephony systems, leading to the GSM standard [Mouly 1992]. Europeans deployed GSM technology with great success in the early 1990s, and since then GSM has grown to be the 800-pound gorilla of the cellular telephone world, with more than 80% of all cellular subscribers worldwide using GSM.

When people talk about cellular technology, they often classify the technology as belonging to one of several "generations." The earliest generations were designed primarily for voice traffic. First generation (1G) systems were analog FDMA systems designed exclusively for voice-only communication. These 1G systems are almost extinct now, having been replaced by digital 2G systems. The original 2G systems were also designed for voice, but later extended (2.5G) to support data (i.e., Internet) as well as voice service. The 3G systems that currently are being deployed also support voice and data, but with an ever increasing emphasis on data capabilities and higher-speed radio access links.

CASE HISTORY

3G CELLULAR MOBILE VERSUS WIRELESS LANS

Many cellular mobile phone operators are deploying 3G cellular mobile systems with indoor data rates of 2 Mbps and outdoor data rates of 384 kbps and higher. These 3G systems are being deployed in licensed radio-frequency bands, with some operators paying considerable sums to governments for spectrum-use licenses. 3G systems allow users to access the Internet from remote outdoor locations while on the move, in a manner similar to today's cellular phone access. For example, 3G technology permits a user to access road map information while driving a car, or movie theater information while sunbathing on a beach. Nevertheless, one may question the extent to which 3G systems will be used, given their cost and the fact that users may often have simultaneous access to both wireless LANs and 3G:

- The emerging wireless LAN infrastructure may become nearly ubiquitous. IEEE 802.11 wireless LANs, operating at 54 Mbps, are enjoying widespread deployment. Almost all portable computers and smartphones are factory-equipped with 802.11 LAN capabilities. Furthermore, emerging Internet appliances—such as wireless cameras and picture frames—will also have small and low-powered wireless LAN capabilities.
- Wireless LAN base stations can also handle mobile phone appliances. Many phones are already capable of connecting to the cellular phone network or to an IP network either natively or using a Skype-like Voice-over-IP service, thus bypassing the operator's cellular voice and 3G data services.

Of course, many other experts believe that 3G not only will be a major success, but will also dramatically revolutionize the way we work and live. Most likely, both WiFi and 3G will both become prevalent wireless technologies, with roaming wireless devices automatically selecting the access technology that provides the best service at their current physical location.

Cellular Network Architecture, 2G: Voice Connections to the Telephone Network

The term *cellular* refers to the fact that the region covered by a cellular network is partitioned into a number of geographic coverage areas, known as **cells**, shown as hexagons on the left side of Figure 6.18. As with the 802.11WiFi standard we studied in Section 6.3.1, GSM has its own particular nomenclature. Each cell contains a **base transceiver station (BTS)** that transmits signals to and receives signals from the mobile stations in its cell. The coverage area of a cell depends

on many factors, including the transmitting power of the BTS, the transmitting power of the user devices, obstructing buildings in the cell, and the height of base station antennas. Although Figure 6.18 shows each cell containing one base transceiver station residing in the middle of the cell, many systems today place the BTS at corners where three cells intersect, so that a single BTS with directional antennas can service three cells.

The GSM standard for 2G cellular systems uses combined FDM/TDM (radio) for the air interface. Recall from Chapter 1 that, with pure FDM, the channel is partitioned into a number of frequency bands with each band devoted to a call. Also recall from Chapter 1 that, with pure TDM, time is partitioned into frames with each frame further partitioned into slots and each call being assigned the use of a particular slot in the revolving frame. In combined FDM/TDM systems, the channel is partitioned into a number of frequency sub-bands; within each sub-band, time is partitioned into frames and slots. Thus, for a combined FDM/TDM system, if the channel is partitioned into F sub-bands and time is partitioned into T slots, then

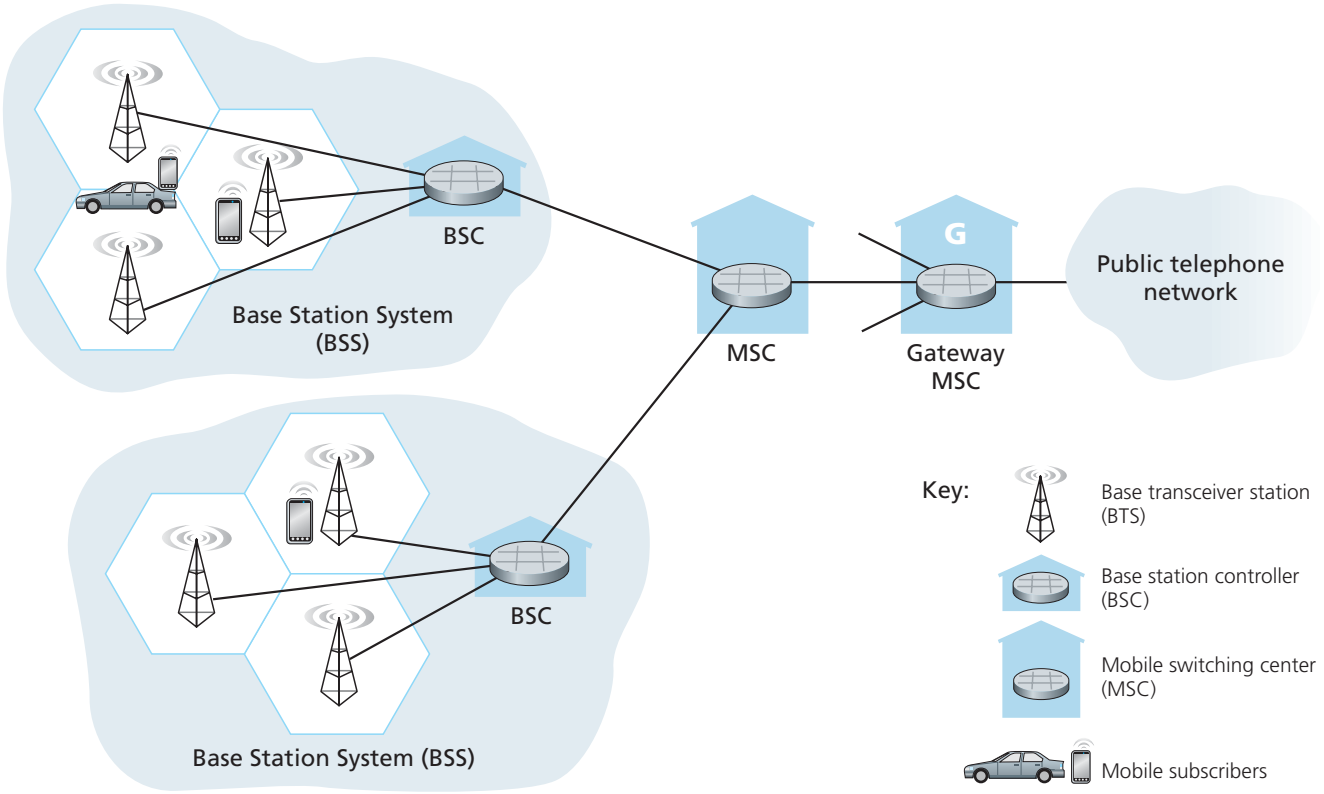


Figure 6.18 ♦ Components of the GSM 2G cellular network architecture

the channel will be able to support $F \cdot T$ simultaneous calls. Recall that we saw in Section 5.3.4 that cable access networks also use a combined FDM/TDM approach. GSM systems consist of 200-kHz frequency bands with each band supporting eight TDM calls. GSM encodes speech at 13 kbps and 12.2 kbps.

A GSM network's **base station controller (BSC)** will typically service several tens of base transceiver stations. The role of the BSC is to allocate BTS radio channels to mobile subscribers, perform **paging** (finding the cell in which a mobile user is resident), and perform handoff of mobile users—a topic we'll cover shortly in Section 6.7.2. The base station controller and its controlled base transceiver stations collectively constitute a GSM **base station system (BSS)**.

As we'll see in Section 6.7, the **mobile switching center (MSC)** plays the central role in user authorization and accounting (e.g., determining whether a mobile device is allowed to connect to the cellular network), call establishment and teardown, and handoff. A single MSC will typically contain up to five BSCs, resulting in approximately 200K subscribers per MSC. A cellular provider's network will have a number of MSCs, with special MSCs known as gateway MSCs connecting the provider's cellular network to the larger public telephone network.

6.4.2 3G Cellular Data Networks: Extending the Internet to Cellular Subscribers

Our discussion in Section 6.4.1 focused on connecting cellular voice users to the public telephone network. But, of course, when we're on the go, we'd also like to read email, access the Web, get location-dependent services (e.g., maps and restaurant recommendations) and perhaps even watch streaming video. To do this, our smartphone will need to run a full TCP/IP protocol stack (including the physical link, network, transport, and application layers) and connect into the Internet via the cellular data network. The topic of cellular data networks is a rather bewildering collection of competing and ever-evolving standards as one generation (and half-generation) succeeds the former and introduces new technologies and services with new acronyms. To make matters worse, there's no single official body that sets requirements for 2.5G, 3G, 3.5G, or 4G technologies, making it hard to sort out the differences among competing standards. In our discussion below, we'll focus on the UMTS (Universal Mobile Telecommunications Service) 3G standards developed by the 3rd Generation Partnership project (3GPP) [3GPP 2012], a widely deployed 3G technology.

Let's take a top-down look at 3G cellular data network architecture shown in Figure 6.19.

3G Core Network

The 3G core cellular data network connects radio access networks to the public Internet. The core network interoperates with components of the existing cellular voice

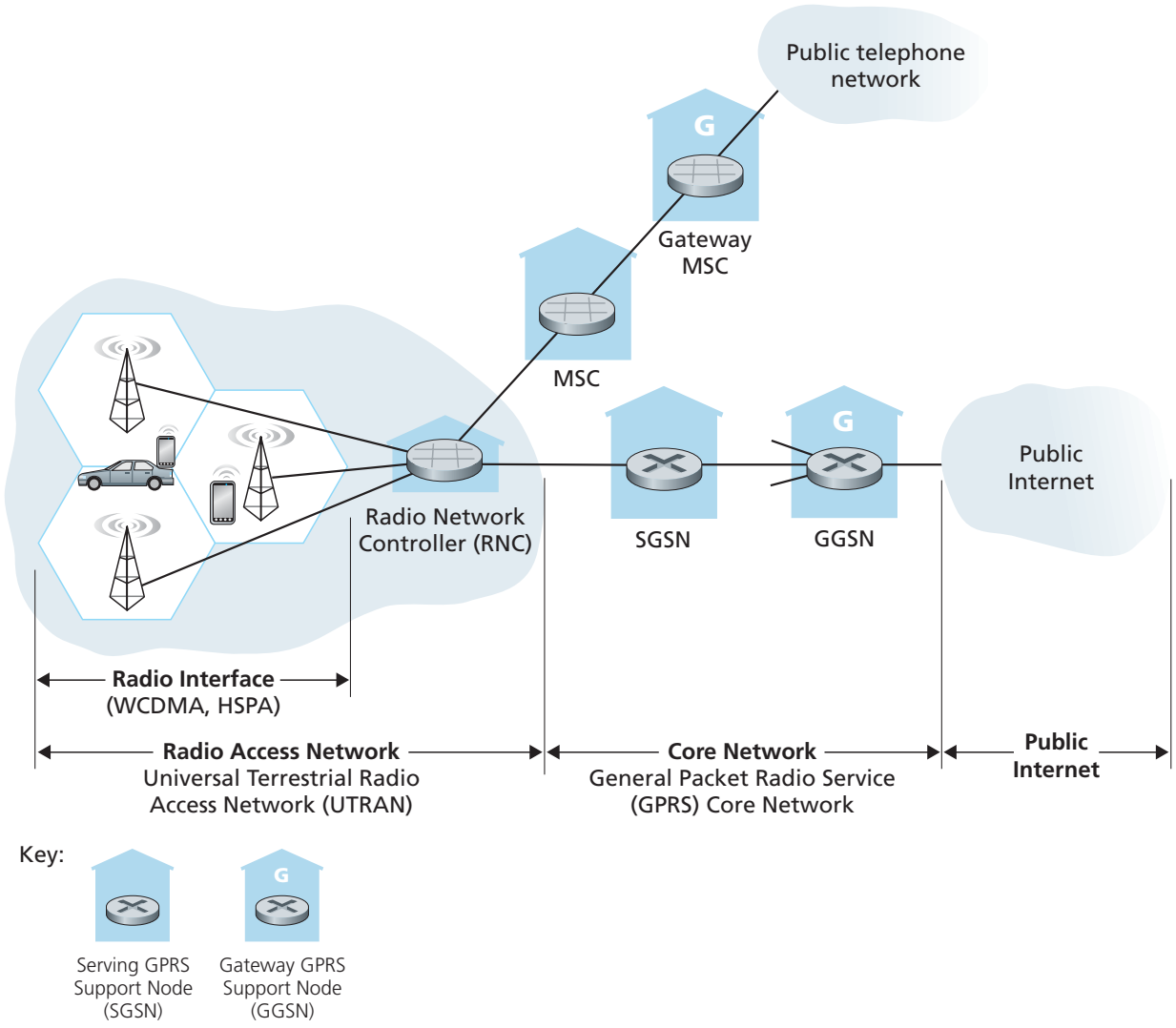


Figure 6.19 ♦ 3G system architecture

network (in particular, the MSC) that we previously encountered in Figure 6.18. Given the considerable amount of existing infrastructure (and profitable services!) in the existing cellular voice network, the approach taken by the designers of 3G data services is clear: *leave the existing core GSM cellular voice network untouched, adding additional cellular data functionality in parallel to the existing cellular voice network*. The alternative—integrating new data services directly into the core of the existing cellular voice network—would have raised the same challenges encountered

in Section 4.4.4, where we discussed integrating new (IPv6) and legacy (IPv4) technologies in the Internet.

There are two types of nodes in the 3G core network: **Serving GPRS Support Nodes (SGSNs)** and **Gateway GPRS Support Nodes (GGSNs)**. (GPRS stands for Generalized Packet Radio Service, an early cellular data service in 2G networks; here we discuss the evolved version of GPRS in 3G networks). An SGSN is responsible for delivering datagrams to/from the mobile nodes in the radio access network to which the SGSN is attached. The SGSN interacts with the cellular voice network's MSC for that area, providing user authorization and handoff, maintaining location (cell) information about active mobile nodes, and performing datagram forwarding between mobile nodes in the radio access network and a GGSN. The GGSN acts as a gateway, connecting multiple SGSNs into the larger Internet. A GGSN is thus the last piece of 3G infrastructure that a datagram originating at a mobile node encounters before entering the larger Internet. To the outside world, the GGSN looks like any other gateway router; the mobility of the 3G nodes within the GGSN's network is hidden from the outside world behind the GGSN.

3G Radio Access Network: The Wireless Edge

The 3G **radio access network** is the wireless first-hop network that we see as a 3G user. The **Radio Network Controller (RNC)** typically controls several cell base transceiver stations similar to the base stations that we encountered in 2G systems (but officially known in 3G UMTS parlance as a "Node Bs"—a rather non-descriptive name!). Each cell's wireless link operates between the mobile nodes and a base transceiver station, just as in 2G networks. The RNC connects to both the circuit-switched cellular voice network via an MSC, and to the packet-switched Internet via an SGSN. Thus, while 3G cellular voice and cellular data services use different core networks, they share a common first/last-hop radio access network.

A significant change in 3G UMTS over 2G networks is that rather than using GSM's FDMA/TDMA scheme, UMTS uses a CDMA technique known as Direct Sequence Wideband CDMA (DS-WCDMA) [Dahlman 1998] within TDMA slots; TDMA slots, in turn, are available on multiple frequencies—an interesting use of all three dedicated channel-sharing approaches that we earlier identified in Chapter 5 and similar to the approach taken in wired cable access networks (see Section 5.3.4). This change requires a new 3G cellular wireless-access network operating in parallel with the 2G BSS radio network shown in Figure 6.19. The data service associated with the WCDMA specification is known as HSP (High Speed Packet Access) and promises downlink data rates of up to 14 Mbps. Details regarding 3G networks can be found at the 3rd Generation Partnership Project (3GPP) Web site [3GPP 2012].

6.4.3 On to 4G: LTE

With 3G systems now being deployed worldwide, can 4G systems be far behind? Certainly not! Indeed, the design, early testing, and initial deployment of 4G systems are already underway. The 4G Long-Term Evolution (LTE) standard put forward by the 3GPP has two important innovations over 3G systems:

- **Evolved Packet Core (EPC)** [3GPP Network Architecture 2012]. The EPC is a simplified all-IP core network that unifies the separate circuit-switched cellular voice network and the packet-switched cellular data network shown in Figure 6.19. It is an "all-IP" network in that both voice and data will be carried in IP datagrams. As we've seen in Chapter 4 and will study in more detail in Chapter 7, IP's "best effort" service model is not inherently well-suited to the stringent performance requirements of Voice-over-IP (VoIP) traffic unless network resources are carefully managed to avoid (rather than react to) congestion. Thus, a key task of the EPC is to manage network resources to provide this high quality of service. The EPC also makes a clear separation between the network control and user data planes, with many of the mobility support features that we will study in Section 6.7 being implemented in the control plane. The EPC allows multiple types of radio access networks, including legacy 2G and 3G radio access networks, to attach to the core network. Two very readable introductions to the EPC are [Motorola 2007; Alcatel-Lucent 2009].
- **LTE Radio Access Network.** LTE uses a combination of frequency division multiplexing and time division multiplexing on the downstream channel, known as orthogonal frequency division multiplexing (OFDM) [Rohde 2008; Ericsson 2011]. (The term "orthogonal" comes from the fact the signals being sent on different frequency channels are created so that they interfere very little with each other, even when channel frequencies are tightly spaced). In LTE, each active mobile node is allocated one or more 0.5 ms time slots in one or more of the channel frequencies. Figure 6.20 shows an allocation of eight time slots over four frequencies. By being allocated increasingly more time slots (whether on the same frequency or on different frequencies), a mobile node is able to achieve increasingly higher transmission rates. Slot (re)allocation among mobile nodes can be performed as often as once every millisecond. Different modulation schemes can also be used to change the transmission rate; see our earlier discussion of Figure 6.3 and dynamic selection of modulation schemes in WiFi networks. Another innovation in the LTE radio network is the use of sophisticated multiple-input, multiple output (MIMO) antennas. The maximum data rate for an LTE user is 100 Mbps in the downstream direction and 50 Mbps in the upstream direction, when using 20 MHz worth of wireless spectrum.

The particular allocation of time slots to mobile nodes is not mandated by the LTE standard. Instead, the decision of which mobile nodes will be allowed to transmit in a given time slot on a given frequency is determined by the scheduling algorithms provided by the LTE equipment vendor and/or the network operator. With opportunistic scheduling [Bender 2000; Kolding 2003; Kulkarni 2005], matching the physical-layer protocol to the channel conditions between the sender and receiver and choosing the receivers to which packets will be sent based on channel conditions allow the radio network controller to make best use of the wireless medium. In addition, user priorities and contracted levels of service (e.g., silver, gold, or platinum) can be used in scheduling downstream packet transmissions. In addition to the LTE capabilities described above, LTE-Advanced allows for downstream bandwidths of hundreds of Mbps by allocating aggregated channels to a mobile node [Akyildiz 2010].

An additional 4G wireless technology—WiMAX (World Interoperability for Microwave Access)—is a family of IEEE 802.16 standards that differ significantly from LTE. Whether LTE or WiMAX becomes the 4G technology of choice is still to be seen, but at the time of this writing (spring 2012), LTE appears to have significantly more momentum. A detailed discussion of WiMAX can be found on this book's Web site.

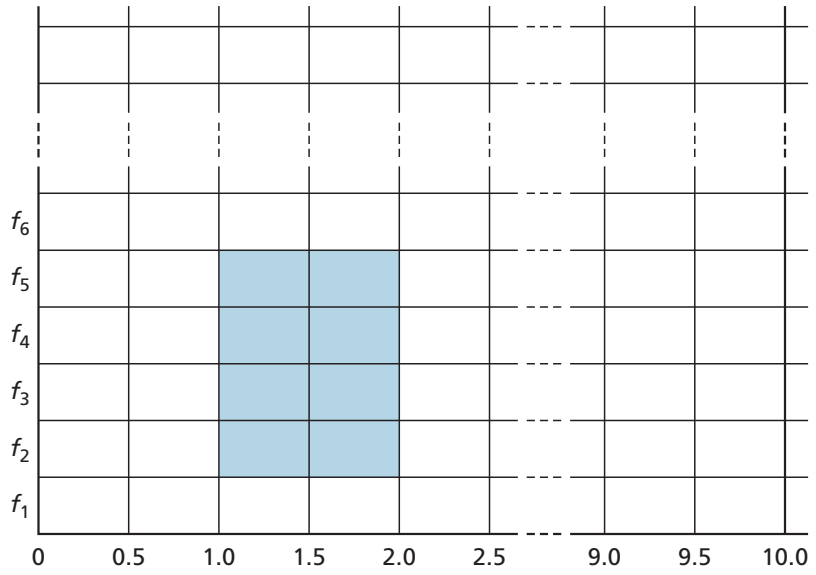


Figure 6.20 ♦ Twenty 0.5 ms slots organized into 10 ms frames at each frequency. An eight-slot allocation is shown shaded.

6.5 Mobility Management: Principles

Having covered the *wireless* nature of the communication links in a wireless network, it's now time to turn our attention to the *mobility* that these wireless links enable. In the broadest sense, a mobile node is one that changes its point of attachment into the network over time. Because the term *mobility* has taken on many meanings in both the computer and telephony worlds, it will serve us well first to consider several dimensions of mobility in some detail.

- *From the network layer's standpoint, how mobile is a user?* A physically mobile user will present a very different set of challenges to the network layer, depending on how he or she moves between points of attachment to the network. At one end of the spectrum in Figure 6.21, a user may carry a laptop with a wireless network interface card around in a building. As we saw in Section 6.3.4, this user is *not* mobile from a network-layer perspective. Moreover, if the user associates with the same access point regardless of location, the user is not even mobile from the perspective of the link layer.

  At the other end of the spectrum, consider the user zooming along the autobahn in a BMW at 150 kilometers per hour, passing through multiple wireless access networks and wanting to maintain an uninterrupted TCP connection to a remote application throughout the trip. This user is *definitely* mobile! In between these extremes is a user who takes a laptop from one location (e.g., office or dormitory) into another (e.g., coffeeshop, classroom) and wants to connect into the network in the new location. This user is also mobile (although less so than the BMW driver!) but does not need to maintain an ongoing connection while moving between points of attachment to the network. Figure 6.21 illustrates this spectrum of user mobility from the network layer's perspective.

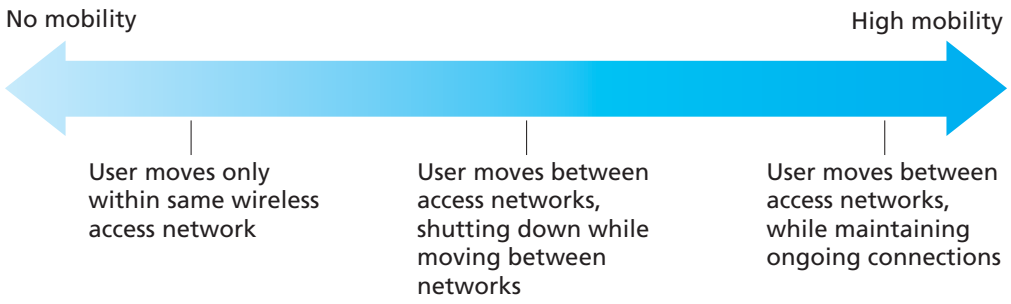


Figure 6.21 ♦ Various degrees of mobility, from the network layer's point of view

- *How important is it for the mobile node's address to always remain the same?* With mobile telephony, your phone number—essentially the network-layer address of your phone—remains the same as you travel from one provider's mobile phone network to another. Must a laptop similarly maintain the same IP address while moving between IP networks?

  The answer to this question will depend strongly on the applications being run. For the BMW driver who wants to maintain an uninterrupted TCP connection to a remote application while zipping along the autobahn, it would be convenient to maintain the same IP address. Recall from Chapter 3 that an Internet application needs to know the IP address and port number of the remote entity with which it is communicating. If a mobile entity is able to maintain its IP address as it moves, mobility becomes invisible from the application standpoint. There is great value to this transparency—an application need not be concerned with a potentially changing IP address, and the same application code serves mobile and nonmobile connections alike. We'll see in the following section that mobile IP provides this transparency, allowing a mobile node to maintain its permanent IP address while moving among networks.

  On the other hand, a less glamorous mobile user might simply want to turn off an office laptop, bring that laptop home, power up, and work from home. If the laptop functions primarily as a client in client-server applications (e.g., send/read e-mail, browse the Web, Telnet to a remote host) from home, the particular IP address used by the laptop is not that important. In particular, one could get by fine with an address that is temporarily allocated to the laptop by the ISP serving the home. We saw in Section 4.4 that DHCP already provides this functionality.
- *What supporting wired infrastructure is available?* In all of our scenarios above, we've implicitly assumed that there is a fixed infrastructure to which the mobile user can connect—for example, the home's ISP network, the wireless access network in the office, or the wireless access networks lining the autobahn. What if no such infrastructure exists? If two users are within communication proximity of each other, can they establish a network connection in the absence of any other network-layer infrastructure? Ad hoc networking provides precisely these capabilities. This rapidly developing area is at the cutting edge of mobile networking research and is beyond the scope of this book. [Perkins 2000] and the IETF Mobile Ad Hoc Network (manet) working group Web pages [manet 2012] provide thorough treatments of the subject.

In order to illustrate the issues involved in allowing a mobile user to maintain ongoing connections while moving between networks, let's consider a human analogy. A twenty-something adult moving out of the family home becomes mobile, living in a series of dormitories and/or apartments, and often changing addresses. If an old friend wants to get in touch, how can that friend find the address of her mobile friend? One common way is to contact the family, since a

mobile adult will often register his or her current address with the family (if for no other reason than so that the parents can send money to help pay the rent!). The family home, with its permanent address, becomes that one place that others can go as a first step in communicating with the mobile adult. Later communication from the friend may be either indirect (for example, with mail being sent first to the parents' home and then forwarded to the mobile adult) or direct (for example, with the friend using the address obtained from the parents to send mail directly to her mobile friend).

In a network setting, the permanent home of a mobile node (such as a laptop or smartphone) is known as the **home network**, and the entity within the home network that performs the mobility management functions discussed below on behalf of the mobile node is known as the **home agent**. The network in which the mobile node is currently residing is known as the **foreign** (or **visited**) **network**, and the entity within the foreign network that helps the mobile node with the mobility management functions discussed below is known as a **foreign agent**. For mobile professionals, their home network might likely be their company network, while the visited network might be the network of a colleague they are visiting. A **correspondent** is the entity wishing to communicate with the mobile node. Figure 6.22 illustrates these concepts, as well as addressing concepts considered below. In Figure 6.22, note that agents are shown as being collocated with routers (e.g., as processes running on routers), but alternatively they could be executing on other hosts or servers in the network.

6.5.1 Addressing

We noted above that in order for user mobility to be transparent to network applications, it is desirable for a mobile node to keep its address as it moves from one network to another. When a mobile node is resident in a foreign network, all traffic addressed to the node's permanent address now needs to be routed to the foreign network. How can this be done? One option is for the foreign network to advertise to all other networks that the mobile node is resident in its network. This could be via the usual exchange of intradomain and interdomain routing information and would require few changes to the existing routing infrastructure. The foreign network could simply advertise to its neighbors that it has a highly specific route to the mobile node's permanent address (that is, essentially inform other networks that it has the correct path for routing datagrams to the mobile node's permanent address; see Section 4.4). These neighbors would then propagate this routing information throughout the network as part of the normal procedure of updating routing information and forwarding tables. When the mobile node leaves one foreign network and joins another, the new foreign network would advertise a new, highly specific route to the mobile node, and the old foreign network would withdraw its routing information regarding the mobile node.

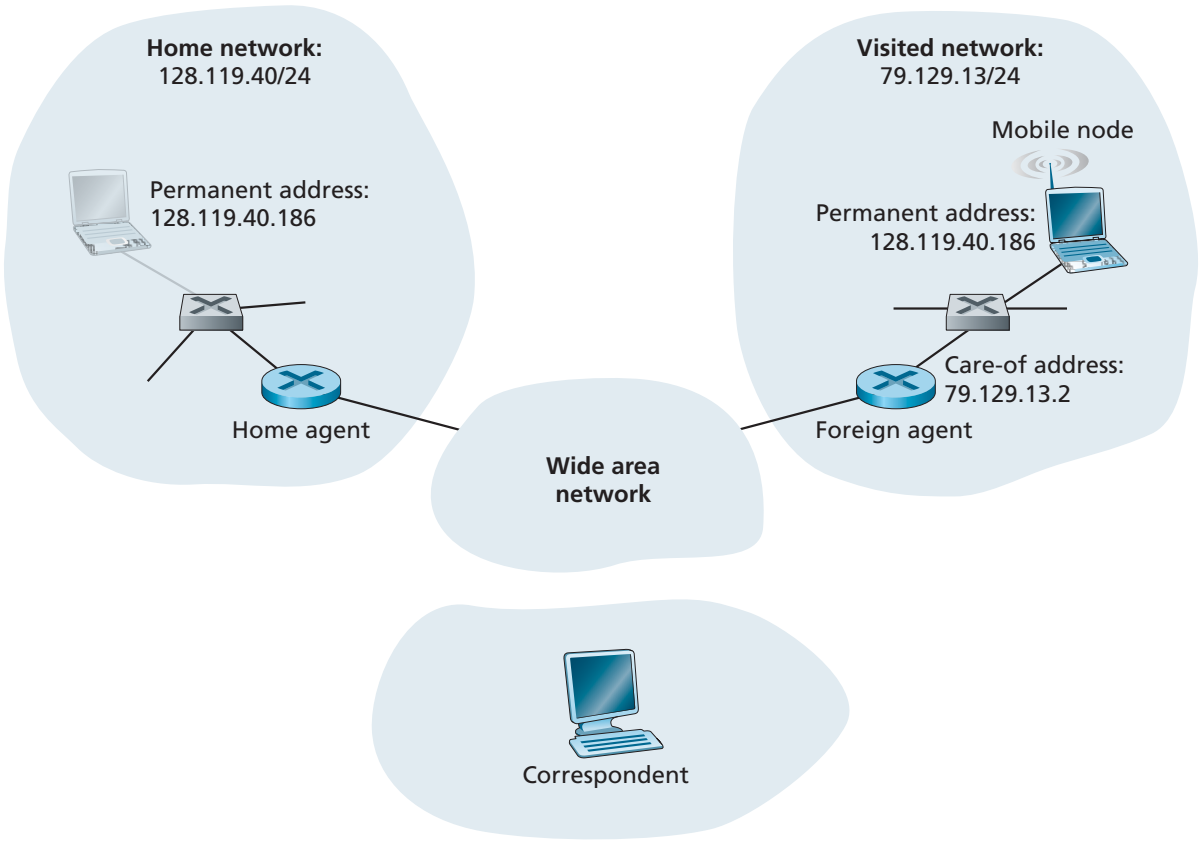


Figure 6.22 ♦ Initial elements of a mobile network architecture

This solves two problems at once, and it does so without making significant changes to the network-layer infrastructure. Other networks know the location of the mobile node, and it is easy to route datagrams to the mobile node, since the forwarding tables will direct datagrams to the foreign network. A significant drawback, however, is that of scalability. If mobility management were to be the responsibility of network routers, the routers would have to maintain forwarding table entries for potentially millions of mobile nodes, and update these entries as nodes move. Some additional drawbacks are explored in the problems at the end of this chapter.

An alternative approach (and one that has been adopted in practice) is to push mobility functionality from the network core to the network edge—a recurring theme in our study of Internet architecture. A natural way to do this is via the mobile node's home network. In much the same way that parents of the mobile twenty-something track their child's location, the home agent in the mobile node's home network can track the foreign network in which the mobile node resides. A protocol

between the mobile node (or a foreign agent representing the mobile node) and the home agent will certainly be needed to update the mobile node's location.

Let's now consider the foreign agent in more detail. The conceptually simplest approach, shown in Figure 6.22, is to locate foreign agents at the edge routers in the foreign network. One role of the foreign agent is to create a so-called **care-of address (COA)** for the mobile node, with the network portion of the COA matching that of the foreign network. There are thus two addresses associated with a mobile node, its **permanent address** (analogous to our mobile youth's family's home address) and its COA, sometimes known as a **foreign address** (analogous to the address of the house in which our mobile youth is currently residing). In the example in Figure 6.22, the permanent address of the mobile node is 128.119.40.186. When visiting network 79.129.13/24, the mobile node has a COA of 79.129.13.2. A second role of the foreign agent is to inform the home agent that the mobile node is resident in its (the foreign agent's) network and has the given COA. We'll see shortly that the COA will be used to "reroute" datagrams to the mobile node via its foreign agent.

Although we have separated the functionality of the mobile node and the foreign agent, it is worth noting that the mobile node can also assume the responsibilities of the foreign agent. For example, the mobile node could obtain a COA in the foreign network (for example, using a protocol such as DHCP) and itself inform the home agent of its COA.

6.5.2 Routing to a Mobile Node

We have now seen how a mobile node obtains a COA and how the home agent can be informed of that address. But having the home agent know the COA solves only part of the problem. How should datagrams be addressed and forwarded to the mobile node? Since only the home agent (and not network-wide routers) knows the location of the mobile node, it will no longer suffice to simply address a datagram to the mobile node's permanent address and send it into the network-layer infrastructure. Something more must be done. Two approaches can be identified, which we will refer to as indirect and direct routing.

Indirect Routing to a Mobile Node

Let's first consider a correspondent that wants to send a datagram to a mobile node. In the **indirect routing** approach, the correspondent simply addresses the datagram to the mobile node's permanent address and sends the datagram into the network, blissfully unaware of whether the mobile node is resident in its home network or is visiting a foreign network; mobility is thus completely transparent to the correspondent. Such datagrams are first routed, as usual, to the mobile node's home network. This is illustrated in step 1 in Figure 6.23.

Let's now turn our attention to the home agent. In addition to being responsible for interacting with a foreign agent to track the mobile node's COA, the home agent

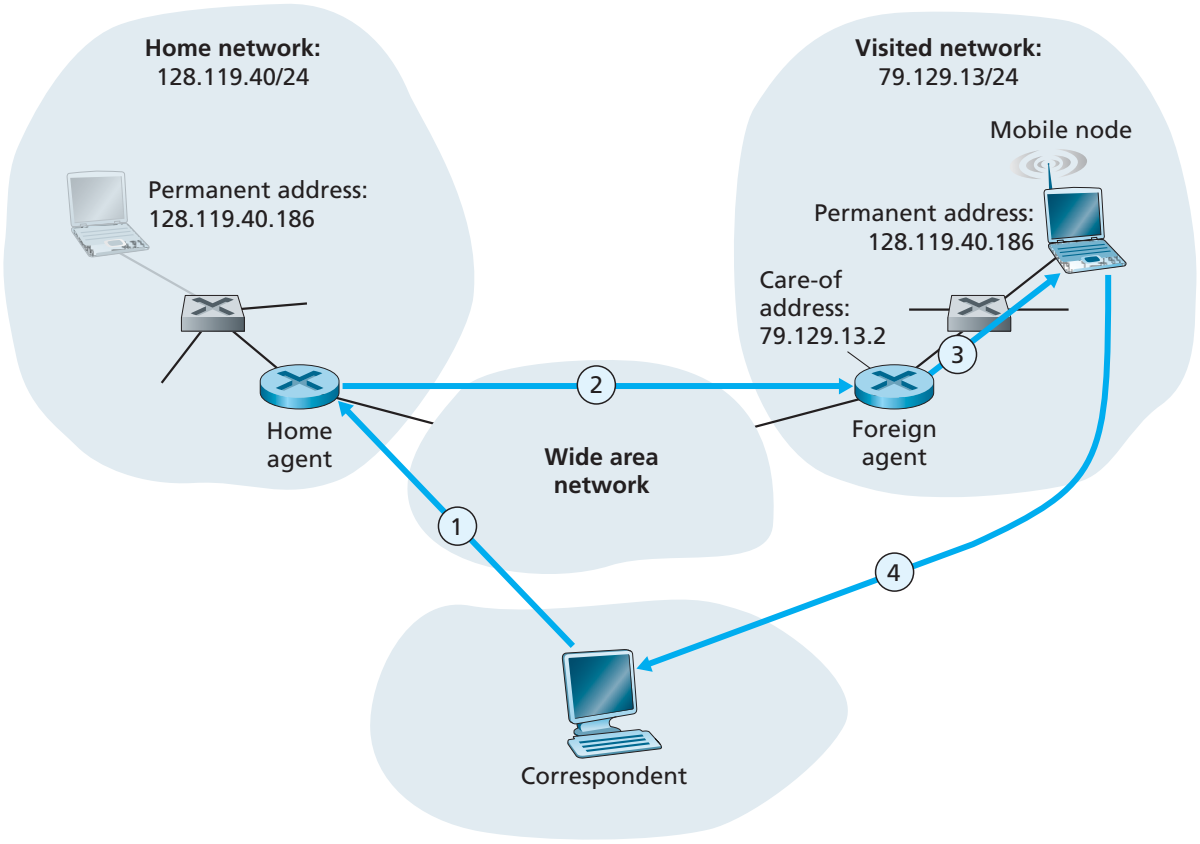


Figure 6.23 ♦ Indirect routing to a mobile node

has another very important function. Its second job is to be on the lookout for arriving datagrams addressed to nodes whose home network is that of the home agent but that are currently resident in a foreign network. The home agent intercepts these datagrams and then forwards them to a mobile node in a two-step process. The datagram is first forwarded to the foreign agent, using the mobile node's COA (step 2 in Figure 6.23), and then forwarded from the foreign agent to the mobile node (step 3 in Figure 6.23).

It is instructive to consider this rerouting in more detail. The home agent will need to address the datagram using the mobile node's COA, so that the network layer will route the datagram to the foreign network. On the other hand, it is desirable to leave the correspondent's datagram intact, since the application receiving the datagram should be unaware that the datagram was forwarded via the home agent. Both goals can be satisfied by having the home agent **encapsulate** the correspondent's original complete datagram within a new (larger) datagram. This larger

datagram is addressed and delivered to the mobile node's COA. The foreign agent, who "owns" the COA, will receive and decapsulate the datagram—that is, remove the correspondent's original datagram from within the larger encapsulating datagram and forward (step 3 in Figure 6.23) the original datagram to the mobile node. Figure 6.24 shows a correspondent's original datagram being sent to the home network, an encapsulated datagram being sent to the foreign agent, and the original datagram being delivered to the mobile node. The sharp reader will note that the encapsulation/decapsulation described here is identical to the notion of tunneling, discussed in Chapter 4 in the context of IP multicast and IPv6.

Let's next consider how a mobile node sends datagrams to a correspondent. This is quite simple, as the mobile node can address its datagram *directly* to the correspondent (using its own permanent address as the source address, and the correspondent's address as the destination address). Since the mobile node knows the correspondent's address, there is no need to route the datagram back through the home agent. This is shown as step 4 in Figure 6.23.

Let's summarize our discussion of indirect routing by listing the new network-layer functionality required to support mobility.

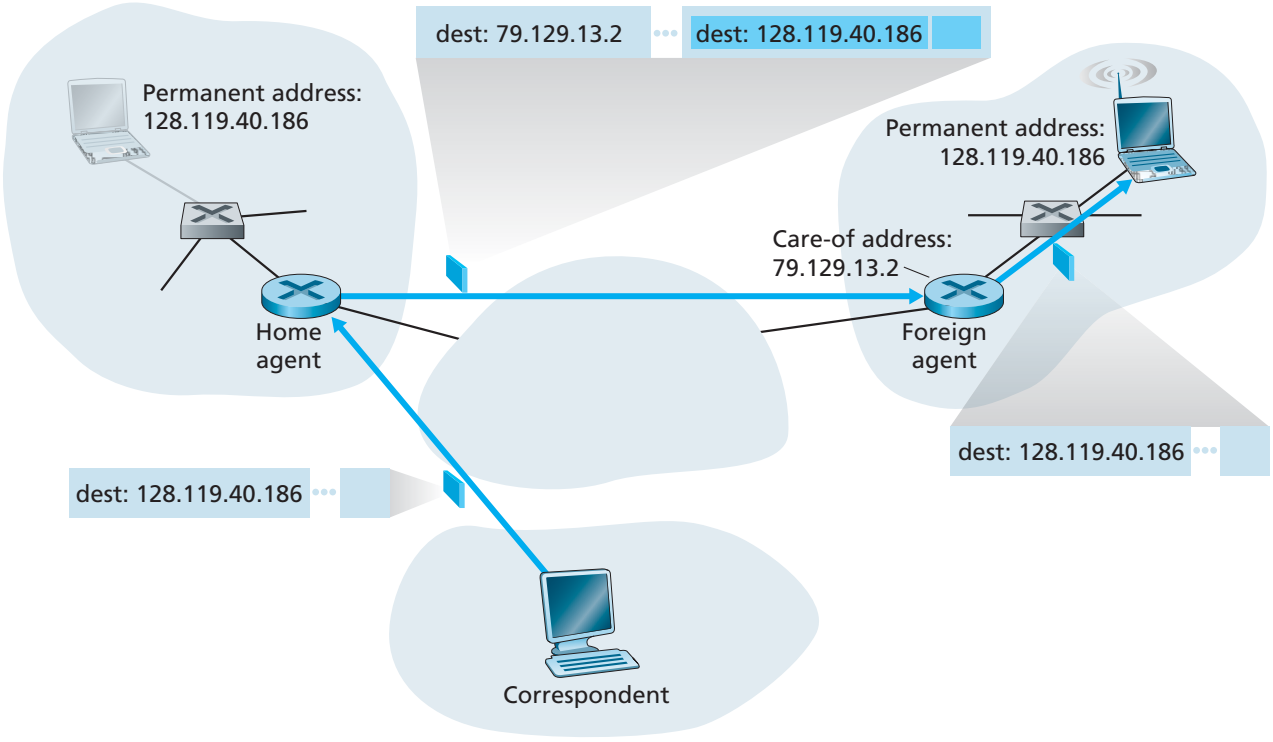


Figure 6.24 ♦ Encapsulation and decapsulation

- *A mobile-node–to–foreign-agent protocol.* The mobile node will register with the foreign agent when attaching to the foreign network. Similarly, a mobile node will deregister with the foreign agent when it leaves the foreign network.
- *A foreign-agent–to–home-agent registration protocol.* The foreign agent will register the mobile node's COA with the home agent. A foreign agent need not explicitly deregister a COA when a mobile node leaves its network, because the subsequent registration of a new COA, when the mobile node moves to a new network, will take care of this.
- *A home-agent datagram encapsulation protocol.* Encapsulation and forwarding of the correspondent's original datagram within a datagram addressed to the COA.
- *A foreign-agent decapsulation protocol.* Extraction of the correspondent's original datagram from the encapsulating datagram, and the forwarding of the original datagram to the mobile node.

The previous discussion provides all the pieces—foreign agents, the home agent, and indirect forwarding—needed for a mobile node to maintain an ongoing connection while moving among networks. As an example of how these pieces fit together, assume the mobile node is attached to foreign network A, has registered a COA in network A with its home agent, and is receiving datagrams that are being indirectly routed through its home agent. The mobile node now moves to foreign network B and registers with the foreign agent in network B, which informs the home agent of the mobile node's new COA. From this point on, the home agent will reroute datagrams to foreign network B. As far as a correspondent is concerned, mobility is transparent—datagrams are routed via the same home agent both before and after the move. As far as the home agent is concerned, there is no disruption in the flow of datagrams—arriving datagrams are first forwarded to foreign network A; after the change in COA, datagrams are forwarded to foreign network B. But will the mobile node see an interrupted flow of datagrams as it moves between networks? As long as the time between the mobile node's disconnection from network A (at which point it can no longer receive datagrams via A) and its attachment to network B (at which point it will register a new COA with its home agent) is small, few datagrams will be lost. Recall from Chapter 3 that end-to-end connections can suffer datagram loss due to network congestion. Hence occasional datagram loss within a connection when a node moves between networks is by no means a catastrophic problem. If loss-free communication is required, upper-layer mechanisms will recover from datagram loss, whether such loss results from network congestion or from user mobility.

An indirect routing approach is used in the mobile IP standard [RFC 5944], as discussed in Section 6.6.

Direct Routing to a Mobile Node

The indirect routing approach illustrated in Figure 6.23 suffers from an inefficiency known as the **triangle routing problem**—datagrams addressed to the mobile node must be routed first to the home agent and then to the foreign network, even when a much more efficient route exists between the correspondent and the mobile node. In the worst case, imagine a mobile user who is visiting the foreign network of a colleague. The two are sitting side by side and exchanging data over the network. Datagrams from the correspondent (in this case the colleague of the visitor) are routed to the mobile user's home agent and then back again to the foreign network!

Direct routing overcomes the inefficiency of triangle routing, but does so at the cost of additional complexity. In the direct routing approach, a **correspondent agent** in the correspondent's network first learns the COA of the mobile node. This can be done by having the correspondent agent query the home agent, assuming that (as in the case of indirect routing) the mobile node has an up-to-date value for its COA registered with its home agent. It is also possible for the correspondent itself to perform the function of the correspondent agent, just as a mobile node could perform the function of the foreign agent. This is shown as steps 1 and 2 in Figure 6.25. The correspondent agent then tunnels datagrams directly to the mobile node's COA, in a manner analogous to the tunneling performed by the home agent, steps 3 and 4 in Figure 6.25.

While direct routing overcomes the triangle routing problem, it introduces two important additional challenges:

- A **mobile-user location protocol** is needed for the correspondent agent to query the home agent to obtain the mobile node's COA (steps 1 and 2 in Figure 6.25).
- When the mobile node moves from one foreign network to another, how will data now be forwarded to the new foreign network? In the case of indirect routing, this problem was easily solved by updating the COA maintained by the home agent. However, with direct routing, the home agent is queried for the COA by the correspondent agent only once, at the beginning of the session. Thus, updating the COA at the home agent, while necessary, will not be enough to solve the problem of routing data to the mobile node's new foreign network.

One solution would be to create a new protocol to notify the correspondent of the changing COA. An alternate solution, and one that we'll see adopted in practice in GSM networks, works as follows. Suppose data is currently being forwarded to the mobile node in the foreign network where the mobile node was located when the session first started (step 1 in Figure 6.26). We'll identify the foreign agent in that foreign network where the mobile node was first found as the **anchor foreign agent**. When the mobile node moves to a new foreign network (step 2 in Figure 6.26), the mobile node registers with the new foreign agent (step 3), and the new foreign agent provides the anchor foreign agent with the mobile node's new COA (step 4). When

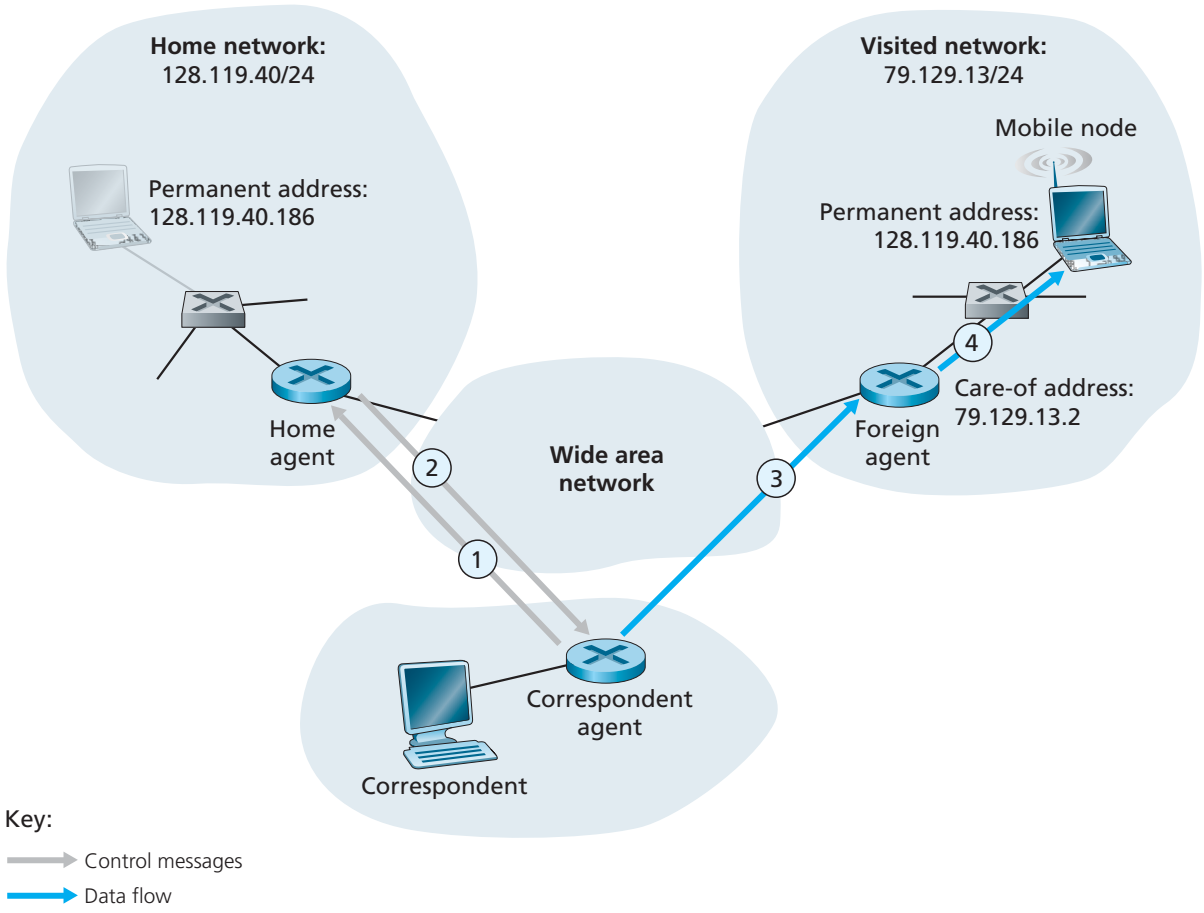


Figure 6.25 ♦ Direct routing to a mobile user

the anchor foreign agent receives an encapsulated datagram for a departed mobile node, it can then re-encapsulate the datagram and forward it to the mobile node (step 5) using the new COA. If the mobile node later moves yet again to a new foreign network, the foreign agent in that new visited network would then contact the anchor foreign agent in order to set up forwarding to this new foreign network.

6.6 Mobile IP

The Internet architecture and protocols for supporting mobility, collectively known as mobile IP, are defined primarily in RFC 5944 for IPv4. Mobile IP is a flexible standard, supporting many different modes of operation (for example, operation

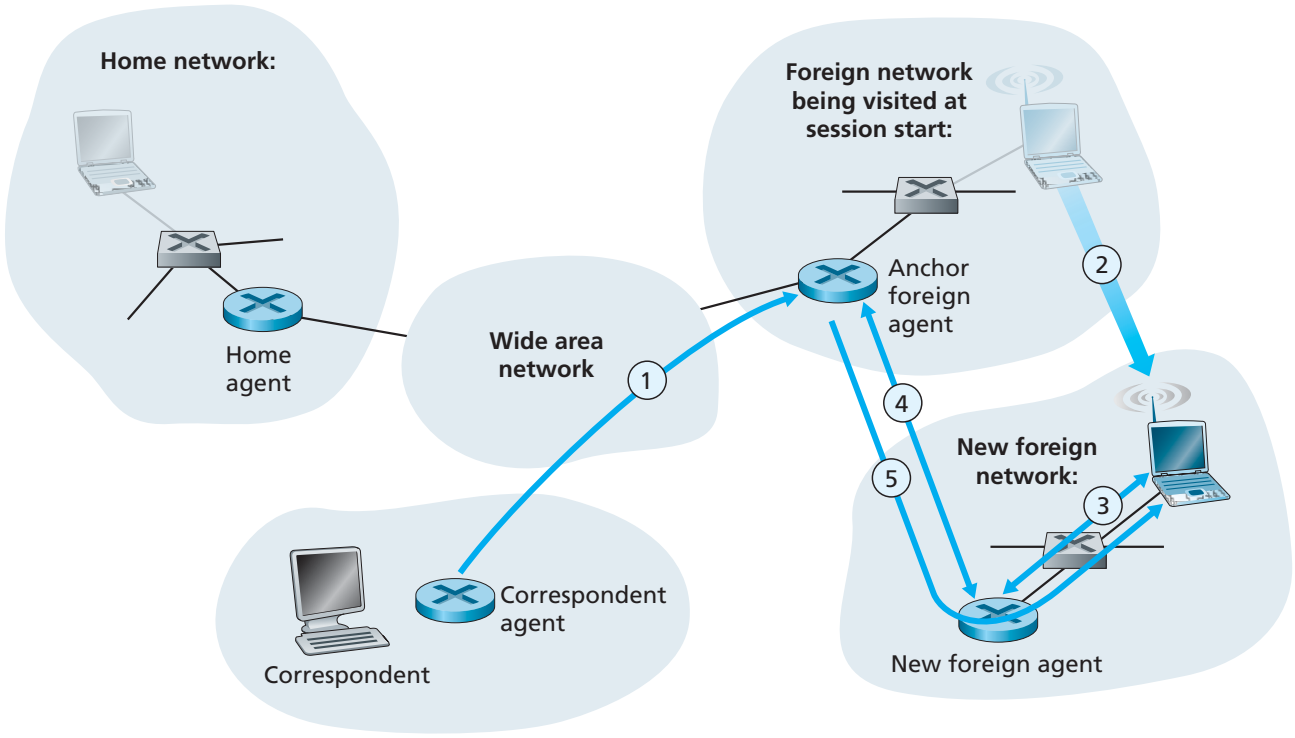


Figure 6.26 ♦ Mobile transfer between networks with direct routing

with or without a foreign agent), multiple ways for agents and mobile nodes to discover each other, use of single or multiple COAs, and multiple forms of encapsulation. As such, mobile IP is a complex standard, and would require an entire book to describe in detail; indeed one such book is [Perkins 1998b]. Our modest goal here is to provide an overview of the most important aspects of mobile IP and to illustrate its use in a few common-case scenarios.

The mobile IP architecture contains many of the elements we have considered above, including the concepts of home agents, foreign agents, care-of addresses, and encapsulation/decapsulation. The current standard [RFC 5944] specifies the use of indirect routing to the mobile node.

The mobile IP standard consists of three main pieces:

- *Agent discovery.* Mobile IP defines the protocols used by a home or foreign agent to advertise its services to mobile nodes, and protocols for mobile nodes to solicit the services of a foreign or home agent.

- *Registration with the home agent.* Mobile IP defines the protocols used by the mobile node and/or foreign agent to register and deregister COAs with a mobile node's home agent.
- *Indirect routing of datagrams.* The standard also defines the manner in which datagrams are forwarded to mobile nodes by a home agent, including rules for forwarding datagrams, rules for handling error conditions, and several forms of encapsulation [RFC 2003, RFC 2004].

Security considerations are prominent throughout the mobile IP standard. For example, authentication of a mobile node is clearly needed to ensure that a malicious user does not register a bogus care-of address with a home agent, which could cause all datagrams addressed to an IP address to be redirected to the malicious user. Mobile IP achieves security using many of the mechanisms that we will examine in Chapter 8, so we will not address security considerations in our discussion below.

Agent Discovery

A mobile IP node arriving to a new network, whether attaching to a foreign network or returning to its home network, must learn the identity of the corresponding foreign or home agent. Indeed it is the discovery of a new foreign agent, with a new network address, that allows the network layer in a mobile node to learn that it has moved into a new foreign network. This process is known as **agent discovery**. Agent discovery can be accomplished in one of two ways: via agent advertisement or via agent solicitation.

With **agent advertisement**, a foreign or home agent advertises its services using an extension to the existing router discovery protocol [RFC 1256]. The agent periodically broadcasts an ICMP message with a type field of 9 (router discovery) on all links to which it is connected. The router discovery message contains the IP address of the router (that is, the agent), thus allowing a mobile node to learn the agent's IP address. The router discovery message also contains a mobility agent advertisement extension that contains additional information needed by the mobile node. Among the more important fields in the extension are the following:

- *Home agent bit (H).* Indicates that the agent is a home agent for the network in which it resides.
- *Foreign agent bit (F).* Indicates that the agent is a foreign agent for the network in which it resides.
- *Registration required bit (R).* Indicates that a mobile user in this network *must* register with a foreign agent. In particular, a mobile user cannot obtain a care-of address in the foreign network (for example, using DHCP) and assume the

functionality of the foreign agent for itself, without registering with the foreign agent.

- *M, G encapsulation bits.* Indicate whether a form of encapsulation other than IP-in-IP encapsulation will be used.
- *Care-of address (COA) fields.* A list of one or more care-of addresses provided by the foreign agent. In our example below, the COA will be associated with the foreign agent, who will receive datagrams sent to the COA and then forward them to the appropriate mobile node. The mobile user will select one of these addresses as its COA when registering with its home agent.

Figure 6.27 illustrates some of the key fields in the agent advertisement message.

With **agent solicitation**, a mobile node wanting to learn about agents without waiting to receive an agent advertisement can broadcast an agent solicitation message, which is simply an ICMP message with type value 10. An agent receiving the solicitation will unicast an agent advertisement directly to the mobile node, which can then proceed as if it had received an unsolicited advertisement.

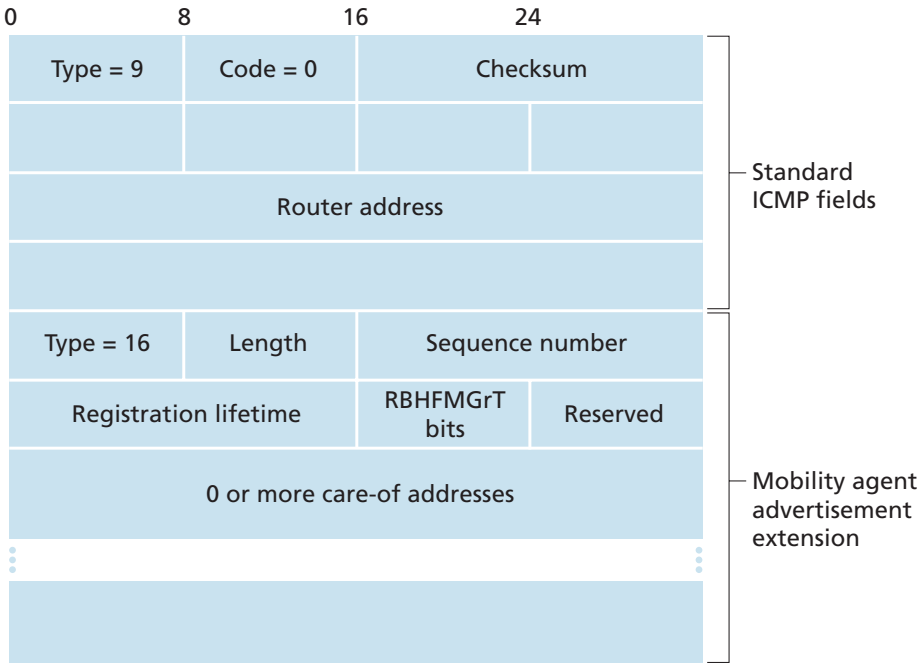


Figure 6.27 ♦ ICMP router discovery message with mobility agent advertisement extension

Registration with the Home Agent

Once a mobile IP node has received a COA, that address must be registered with the home agent. This can be done either via the foreign agent (who then registers the COA with the home agent) or directly by the mobile IP node itself. We consider the former case below. Four steps are involved.

1. Following the receipt of a foreign agent advertisement, a mobile node sends a mobile IP registration message to the foreign agent. The registration message is carried within a UDP datagram and sent to port 434. The registration message carries a COA advertised by the foreign agent, the address of the home agent (HA), the permanent address of the mobile node (MA), the requested lifetime of the registration, and a 64-bit registration identification. The requested registration lifetime is the number of seconds that the registration is to be valid. If the registration is not renewed at the home agent within the specified lifetime, the registration will become invalid. The registration identifier acts like a sequence number and serves to match a received registration reply with a registration request, as discussed below.
2. The foreign agent receives the registration message and records the mobile node's permanent IP address. The foreign agent now knows that it should be looking for datagrams containing an encapsulated datagram whose destination address matches the permanent address of the mobile node. The foreign agent then sends a mobile IP registration message (again, within a UDP datagram) to port 434 of the home agent. The message contains the COA, HA, MA, encapsulation format requested, requested registration lifetime, and registration identification.
3. The home agent receives the registration request and checks for authenticity and correctness. The home agent binds the mobile node's permanent IP address with the COA; in the future, datagrams arriving at the home agent and addressed to the mobile node will now be encapsulated and tunneled to the COA. The home agent sends a mobile IP registration reply containing the HA, MA, actual registration lifetime, and the registration identification of the request that is being satisfied with this reply.
4. The foreign agent receives the registration reply and then forwards it to the mobile node.

At this point, registration is complete, and the mobile node can receive datagrams sent to its permanent address. Figure 6.28 illustrates these steps. Note that the home agent specifies a lifetime that is smaller than the lifetime requested by the mobile node.

A foreign agent need not explicitly deregister a COA when a mobile node leaves its network. This will occur automatically, when the mobile node moves to a new network (whether another foreign network or its home network) and registers a new COA.

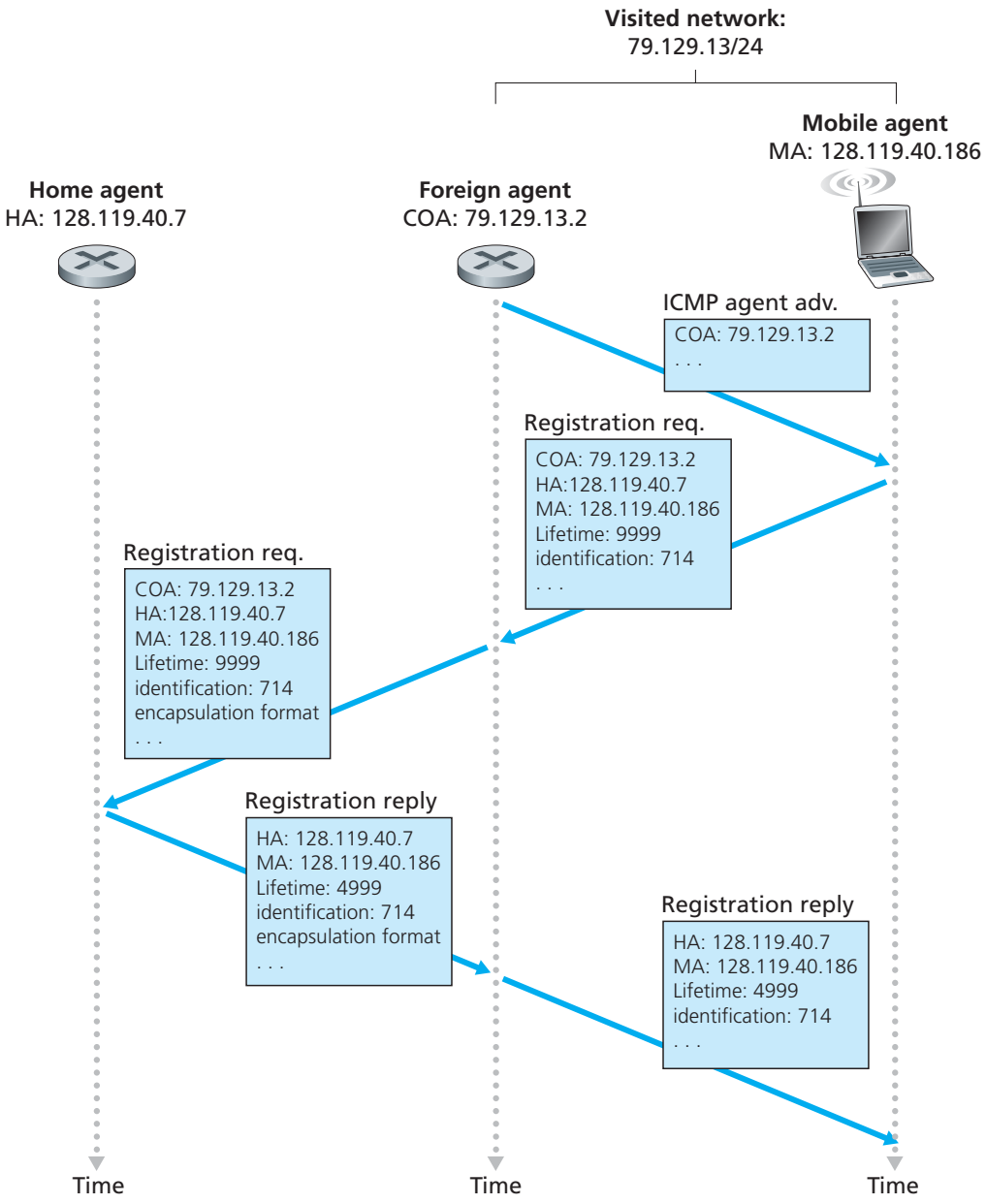


Figure 6.28 ♦ Agent advertisement and mobile IP registration

The mobile IP standard allows many additional scenarios and capabilities in addition to those described previously. The interested reader should consult [Perkins 1998b; RFC 5944].

6.7 Managing Mobility in Cellular Networks

Having examined how mobility is managed in IP networks, let's now turn our attention to networks with an even longer history of supporting mobility—cellular telephony networks. Whereas we focused on the first-hop wireless link in cellular networks in Section 6.4, we'll focus here on mobility, using the GSM cellular network architecture [Goodman 1997; Mouly 1992; Scourias 2012; Kaaranen 2001; Korhonen 2003; Turner 2012] as our case study, since it is a mature and widely deployed technology. As in the case of mobile IP, we'll see that a number of the fundamental principles we identified in Section 6.5 are embodied in GSM's network architecture.

Like mobile IP, GSM adopts an indirect routing approach (see Section 6.5.2), first routing the correspondent's call to the mobile user's home network and from there to the visited network. In GSM terminology, the mobile users's home network is referred to as the mobile user's **home public land mobile network (home PLMN)**. Since the PLMN acronym is a bit of a mouthful, and mindful of our quest to avoid an alphabet soup of acronyms, we'll refer to the GSM home PLMN simply as the **home network**. The home network is the cellular provider with which the mobile user has a subscription (i.e., the provider that bills the user for monthly cellular service). The visited PLMN, which we'll refer to simply as the **visited network**, is the network in which the mobile user is currently residing.

As in the case of mobile IP, the responsibilities of the home and visited networks are quite different.

- The home network maintains a database known as the **home location register (HLR)**, which contains the permanent cell phone number and subscriber profile information for each of its subscribers. Importantly, the HLR also contains information about the current locations of these subscribers. That is, if a mobile user is currently roaming in another provider's cellular network, the HLR contains enough information to obtain (via a process we'll describe shortly) an address in the visited network to which a call to the mobile user should be routed. As we'll see, a special switch in the home network, known as the **Gateway Mobile services Switching Center (GMSC)** is contacted by a correspondent when a call is placed to a mobile user. Again, in our quest to avoid an alphabet soup of acronyms, we'll refer to the GMSC here by a more descriptive term, **home MSC**.
- The visited network maintains a database known as the **visitor location register (VLR)**. The VLR contains an entry for each mobile user that is *currently* in the portion of the network served by the VLR. VLR entries thus come and go as mobile users enter and leave the network. A VLR is usually co-located with the mobile switching center (MSC) that coordinates the setup of a call to and from the visited network.

In practice, a provider's cellular network will serve as a home network for its subscribers and as a visited network for mobile users whose subscription is with a different cellular provider.

6.7.1 Routing Calls to a Mobile User

We're now in a position to describe how a call is placed to a mobile GSM user in a visited network. We'll consider a simple example below; more complex scenarios are described in [Mouly 1992]. The steps, as illustrated in Figure 6.29, are as follows:

1. The correspondent dials the mobile user's phone number. This number itself does not refer to a particular telephone line or location (after all, the phone number is fixed and the user is mobile!). The leading digits in the number are sufficient to globally identify the mobile's home network. The call is routed from the correspondent through the PSTN to the home MSC in the mobile's home network. This is the first leg of the call.
2. The home MSC receives the call and interrogates the HLR to determine the location of the mobile user. In the simplest case, the HLR returns the **mobile**

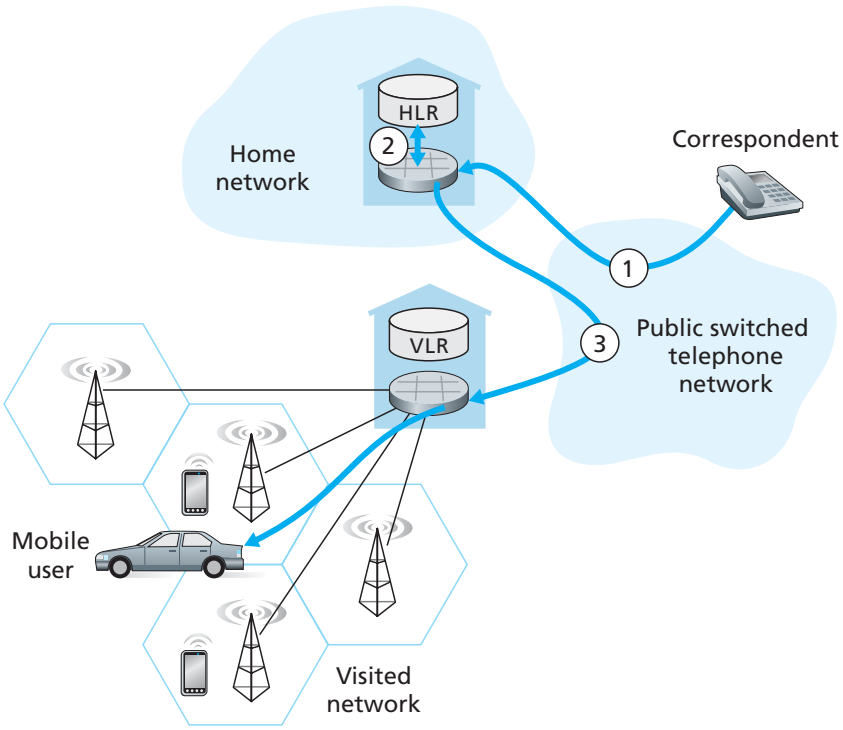


Figure 6.29 ♦ Placing a call to a mobile user: indirect routing

station roaming number (MSRN), which we will refer to as the **roaming number**. Note that this number is different from the mobile's permanent phone number, which is associated with the mobile's home network. The roaming number is ephemeral: It is temporarily assigned to a mobile when it enters a visited network. The roaming number serves a role similar to that of the care-of address in mobile IP and, like the COA, is invisible to the correspondent and the mobile. If HLR does not have the roaming number, it returns the address of the VLR in the visited network. In this case (not shown in Figure 6.29), the home MSC will need to query the VLR to obtain the roaming number of the mobile node. But how does the HLR get the roaming number or the VLR address in the first place? What happens to these values when the mobile user moves to another visited network? We'll consider these important questions shortly.

3. Given the roaming number, the home MSC sets up the second leg of the call through the network to the MSC in the visited network. The call is completed, being routed from the correspondent to the home MSC, and from there to the visited MSC, and from there to the base station serving the mobile user.

An unresolved question in step 2 is how the HLR obtains information about the location of the mobile user. When a mobile telephone is switched on or enters a part of a visited network that is covered by a new VLR, the mobile must register with the visited network. This is done through the exchange of signaling messages between the mobile and the VLR. The visited VLR, in turn, sends a location update request message to the mobile's HLR. This message informs the HLR of either the roaming number at which the mobile can be contacted, or the address of the VLR (which can then later be queried to obtain the mobile number). As part of this exchange, the VLR also obtains subscriber information from the HLR about the mobile and determines what services (if any) should be accorded the mobile user by the visited network.

6.7.2 Handoffs in GSM

A **handoff** occurs when a mobile station changes its association from one base station to another during a call. As shown in Figure 6.30, a mobile's call is initially (before handoff) routed to the mobile through one base station (which we'll refer to as the old base station), and after handoff is routed to the mobile through another base station (which we'll refer to as the new base station). Note that a handoff between base stations results not only in the mobile transmitting/receiving to/from a new base station, but also in the rerouting of the ongoing call from a switching point within the network to the new base station. Let's initially assume that the old and new base stations share the same MSC, and that the rerouting occurs at this MSC.

There may be several reasons for handoff to occur, including (1) the signal between the current base station and the mobile may have deteriorated to such an extent that the call is in danger of being dropped, and (2) a cell may have become

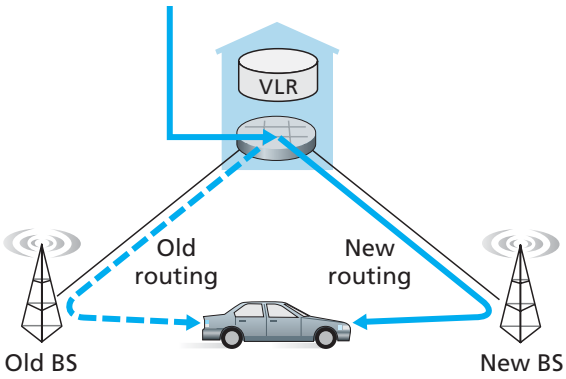


Figure 6.30 ♦ Handoff scenario between base stations with a common MSC

overloaded, handling a large number of calls. This congestion may be alleviated by handing off mobiles to less congested nearby cells.

While it is associated with a base station, a mobile periodically measures the strength of a beacon signal from its current base station as well as beacon signals from nearby base stations that it can "hear." These measurements are reported once or twice a second to the mobile's current base station. Handoff in GSM is initiated by the old base station based on these measurements, the current loads of mobiles in nearby cells, and other factors [Mouly 1992]. The GSM standard does not specify the specific algorithm to be used by a base station to determine whether or not to perform handoff.

Figure 6.31 illustrates the steps involved when a base station does decide to hand off a mobile user:

1. The old base station (BS) informs the visited MSC that a handoff is to be performed and the BS (or possible set of BSs) to which the mobile is to be handed off.
2. The visited MSC initiates path setup to the new BS, allocating the resources needed to carry the rerouted call, and signaling the new BS that a handoff is about to occur.
3. The new BS allocates and activates a radio channel for use by the mobile.
4. The new BS signals back to the visited MSC and the old BS that the visited-MSC-to-new-BS path has been established and that the mobile should be informed of the impending handoff. The new BS provides all of the information that the mobile will need to associate with the new BS.
5. The mobile is informed that it should perform a handoff. Note that up until this point, the mobile has been blissfully unaware that the network has been laying the groundwork (e.g., allocating a channel in the new BS and allocating a path from the visited MSC to the new BS) for a handoff.
6. The mobile and the new BS exchange one or more messages to fully activate the new channel in the new BS.

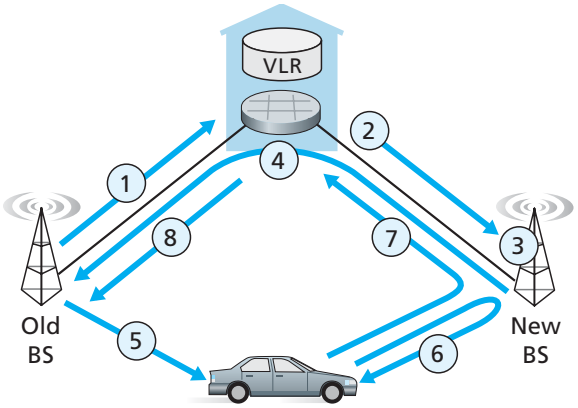


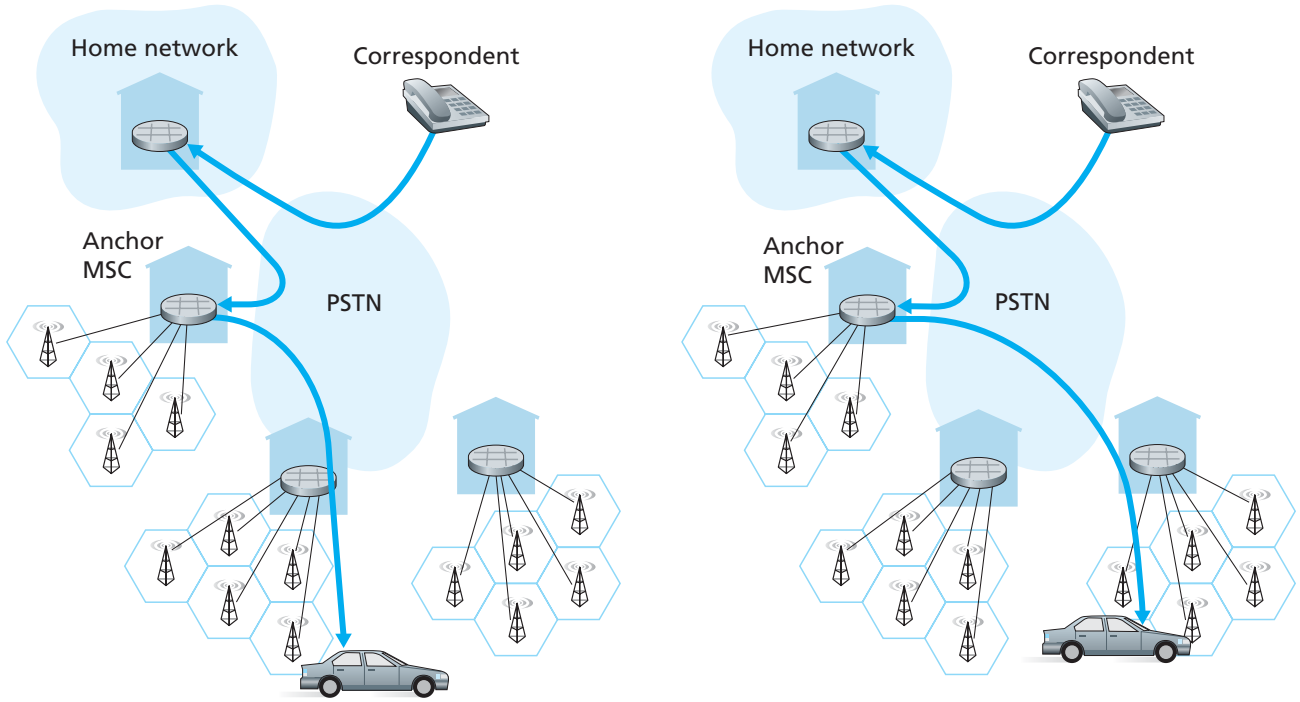
Figure 6.31 ♦ Steps in accomplishing a handoff between base stations with a common MSC

7. The mobile sends a handoff complete message to the new BS, which is forwarded up to the visited MSC. The visited MSC then reroutes the ongoing call to the mobile via the new BS.
8. The resources allocated along the path to the old BS are then released.

Let's conclude our discussion of handoff by considering what happens when the mobile moves to a BS that is associated with a *different* MSC than the old BS, and what happens when this inter-MSC handoff occurs more than once. As shown in Figure 6.32, GSM defines the notion of an **anchor MSC**. The anchor MSC is the MSC visited by the mobile when a call first begins; the anchor MSC thus remains unchanged during the call. Throughout the call's duration and regardless of the number of inter-MSC transfers performed by the mobile, the call is routed from the home MSC to the anchor MSC, and then from the anchor MSC to the visited MSC where the mobile is currently located. When a mobile moves from the coverage area of one MSC to another, the ongoing call is rerouted from the anchor MSC to the new visited MSC containing the new base station. Thus, at all times there are at most three MSCs (the home MSC, the anchor MSC, and the visited MSC) between the correspondent and the mobile. Figure 6.32 illustrates the routing of a call among the MSCs visited by a mobile user.

Rather than maintaining a single MSC hop from the anchor MSC to the current MSC, an alternative approach would have been to simply chain the MSCs visited by the mobile, having an old MSC forward the ongoing call to the new MSC each time the mobile moves to a new MSC. Such MSC chaining can in fact occur in IS-41 cellular networks, with an optional path minimization step to remove MSCs between the anchor MSC and the current visited MSC [Lin 2001].

Let's wrap up our discussion of GSM mobility management with a comparison of mobility management in GSM and Mobile IP. The comparison in Table 6.2 indicates that although IP and cellular networks are fundamentally different in many ways, they share a surprising number of common functional elements and overall approaches in handling mobility.



a. Before handoff

b. After handoff

Figure 6.32 ♦ Rerouting via the anchor MSC

6.8 Wireless and Mobility: Impact on Higher-Layer Protocols

In this chapter, we've seen that wireless networks differ significantly from their wired counterparts at both the link layer (as a result of wireless channel characteristics such as fading, multipath, and hidden terminals) and at the network layer (as a result of mobile users who change their points of attachment to the network). But are there important differences at the transport and application layers? It's tempting to think that these differences will be minor, since the network layer provides the same best-effort delivery service model to upper layers in both wired and wireless networks. Similarly, if protocols such as TCP or UDP are used to provide transport-layer services to applications in both wired and wireless networks, then the application layer should remain unchanged as well. In one sense our intuition is right—TCP and UDP can (and do) operate in networks with wireless links. On the other hand, transport protocols in general, and TCP in particular, can sometimes have very different performance in wired and wireless networks, and it is here, in terms of performance, that differences are manifested. Let's see why.

Recall that TCP retransmits a segment that is either lost or corrupted on the path between sender and receiver. In the case of mobile users, loss can result from either

GSM element	Comment on GSM element	Mobile IP element
Home system	Network to which the mobile user's permanent phone number belongs.	Home network
Gateway mobile switching center or simply home MSC, Home location register (HLR)	Home MSC: point of contact to obtain routable address of mobile user. HLR: database in home system containing permanent phone number, profile information, current location of mobile user, subscription information.	Home agent
Visited system	Network other than home system where mobile user is currently residing.	Visited network.
Visited mobile services switching center, Visitor location register (VLR)	Visited MSC: responsible for setting up calls to/from mobile nodes in cells associated with MSC. VLR: temporary database entry in visited system, containing subscription information for each visiting mobile user.	Foreign agent
Mobile station roaming number (MSRN) or simply roaming number	Routable address for telephone call segment between home MSC and visited MSC, visible to neither the mobile nor the correspondent.	Care-of address

Table 6.2 ♦ Commonalities between mobile IP and GSM mobility

network congestion (router buffer overflow) or from handoff (e.g., from delays in rerouting segments to a mobile's new point of attachment to the network). In all cases, TCP's receiver-to-sender ACK indicates only that a segment was not received intact; the sender is unaware of whether the segment was lost due to congestion, during handoff, or due to detected bit errors. In all cases, the sender's response is the same—to retransmit the segment. TCP's congestion-control response is *also* the same in all cases—TCP decreases its congestion window, as discussed in Section 3.7. By unconditionally decreasing its congestion window, TCP implicitly assumes that segment loss results from congestion rather than corruption or handoff. We saw in Section 6.2 that bit errors are much more common in wireless networks than in wired networks. When such bit errors occur or when handoff loss occurs, there's really no reason for the TCP sender to decrease its congestion window (and thus decrease its sending rate). Indeed, it may well be the case that router buffers are empty and packets are flowing along the end-to-end path unimpeded by congestion.

Researchers realized in the early to mid 1990s that given high bit error rates on wireless links and the possibility of handoff loss, TCP's congestion-control response could be problematic in a wireless setting. Three broad classes of approaches are possible for dealing with this problem:

- *Local recovery.* Local recovery protocols recover from bit errors when and where (e.g., at the wireless link) they occur, e.g., the 802.11 ARQ protocol we studied

in Section 6.3, or more sophisticated approaches that use both ARQ and FEC [Ayanoglu 1995].

- *TCP sender awareness of wireless links.* In the local recovery approaches, the TCP sender is blissfully unaware that its segments are traversing a wireless link. An alternative approach is for the TCP sender and receiver to be aware of the existence of a wireless link, to distinguish between congestive losses occurring in the wired network and corruption/loss occurring at the wireless link, and to invoke congestion control only in response to congestive wired-network losses. [Balakrishnan 1997] investigates various types of TCP, assuming that end systems can make this distinction. [Liu 2003] investigates techniques for distinguishing between losses on the wired and wireless segments of an end-to-end path.
- *Split-connection approaches.* In a split-connection approach [Bakre 1995], the end-to-end connection between the mobile user and the other end point is broken into two transport-layer connections: one from the mobile host to the wireless access point, and one from the wireless access point to the other communication end point (which we'll assume here is a wired host). The end-to-end connection is thus formed by the concatenation of a wireless part and a wired part. The transport layer over the wireless segment can be a standard TCP connection [Bakre 1995], or a specially tailored error recovery protocol on top of UDP. [Yavatkar 1994] investigates the use of a transport-layer selective repeat protocol over the wireless connection. Measurements reported in [Wei 2006] indicate that split TCP connections are widely used in cellular data networks, and that significant improvements can indeed be made through the use of split TCP connections.

Our treatment of TCP over wireless links has been necessarily brief here. In-depth surveys of TCP challenges and solutions in wireless networks can be found in [Hanabali 2005; Leung 2006]. We encourage you to consult the references for details of this ongoing area of research.

Having considered transport-layer protocols, let us next consider the effect of wireless and mobility on application-layer protocols. Here, an important consideration is that wireless links often have relatively low bandwidths, as we saw in Figure 6.2. As a result, applications that operate over wireless links, particularly over cellular wireless links, must treat bandwidth as a scarce commodity. For example, a Web server serving content to a Web browser executing on a 3G phone will likely not be able to provide the same image-rich content that it gives to a browser operating over a wired connection. Although wireless links do provide challenges at the application layer, the mobility they enable also makes possible a rich set of location-aware and context-aware applications [Chen 2000; Baldauf 2007]. More generally, wireless and mobile networks will play a key role in realizing the ubiquitous computing environments of the future [Weiser 1991]. It's fair to say that we've only seen the tip of the iceberg when it comes to the impact of wireless and mobile networks on networked applications and their protocols!

6.9 Summary

Wireless and mobile networks have revolutionized telephony and are having an increasingly profound impact in the world of computer networks as well. With their anytime, anywhere, untethered access into the global network infrastructure, they are not only making network access more ubiquitous, they are also enabling an exciting new set of location-dependent services. Given the growing importance of wireless and mobile networks, this chapter has focused on the principles, common link technologies, and network architectures for supporting wireless and mobile communication.

We began this chapter with an introduction to wireless and mobile networks, drawing an important distinction between the challenges posed by the *wireless* nature of the communication links in such networks, and by the *mobility* that these wireless links enable. This allowed us to better isolate, identify, and master the key concepts in each area. We focused first on wireless communication, considering the characteristics of a wireless link in Section 6.2. In Sections 6.3 and 6.4, we examined the link-level aspects of the IEEE 802.11 (WiFi) wireless LAN standard, two IEEE 802.15 personal area networks (Bluetooth and Zigbee), and 3G and 4G cellular Internet access. We then turned our attention to the issue of mobility. In Section 6.5, we identified several forms of mobility, with points along this spectrum posing different challenges and admitting different solutions. We considered the problems of locating and routing to a mobile user, as well as approaches for handing off the mobile user who dynamically moves from one point of attachment to the network to another. We examined how these issues were addressed in the mobile IP standard and in GSM, in Sections 6.6 and 6.7, respectively. Finally, we considered the impact of wireless links and mobility on transport-layer protocols and networked applications in Section 6.8.

Although we have devoted an entire chapter to the study of wireless and mobile networks, an entire book (or more) would be required to fully explore this exciting and rapidly expanding field. We encourage you to delve more deeply into this field by consulting the many references provided in this chapter.

Homework Problems and Questions

Chapter 6 Review Questions

SECTION 6.1

R1. What does it mean for a wireless network to be operating in "infrastructure mode?" If the network is not in infrastructure mode, what mode of operation is it in, and what is the difference between that mode of operation and infrastructure mode?

R2. What are the four types of wireless networks identified in our taxonomy in Section 6.1? Which of these types of wireless networks have you used?

SECTION 6.2

R3. What are the differences between the following types of wireless channel impairments: path loss, multipath propagation, interference from other sources?

R4. As a mobile node gets farther and farther away from a base station, what are two actions that a base station could take to ensure that the loss probability of a transmitted frame does not increase?

SECTIONS 6.3 AND 6.4

R5. Describe the role of the beacon frames in 802.11.

R6. True or false: Before an 802.11 station transmits a data frame, it must first send an RTS frame and receive a corresponding CTS frame.

R7. Why are acknowledgments used in 802.11 but not in wired Ethernet?

R8. True or false: Ethernet and 802.11 use the same frame structure.

R9. Describe how the RTS threshold works.

R10. Suppose the IEEE 802.11 RTS and CTS frames were as long as the standard DATA and ACK frames. Would there be any advantage to using the CTS and RTS frames? Why or why not?

R11. Section 6.3.4 discusses 802.11 mobility, in which a wireless station moves from one BSS to another within the same subnet. When the APs are interconnected with a switch, an AP may need to send a frame with a spoofed MAC address to get the switch to forward the frame properly. Why?

R12. What are the differences between a master device in a Bluetooth network and a base station in an 802.11 network?

R13. What is meant by a super frame in the 802.15.4 Zigbee standard?

R14. What is the role of the "core network" in the 3G cellular data architecture?

R15. What is the role of the RNC in the 3G cellular data network architecture? What role does the RNC play in the cellular voice network?

SECTIONS 6.5 AND 6.6

R16. If a node has a wireless connection to the Internet, does that node have to be mobile? Explain. Suppose that a user with a laptop walks around her house with her laptop, and always accesses the Internet through the same access point. Is this user mobile from a network standpoint? Explain.

R17. What is the difference between a permanent address and a care-of address? Who assigns a care-of address?

R18. Consider a TCP connection going over Mobile IP. True or false: The TCP connection phase between the correspondent and the mobile host goes through the mobile's home network, but the data transfer phase is directly between the correspondent and the mobile host, bypassing the home network.

SECTION 6.7

R19. What are the purposes of the HLR and VLR in GSM networks? What elements of mobile IP are similar to the HLR and VLR?

R20. What is the role of the anchor MSC in GSM networks?

SECTION 6.8

R21. What are three approaches that can be taken to avoid having a single wireless link degrade the performance of an end-to-end transport-layer TCP connection?

Problems

P1. Consider the single-sender CDMA example in Figure 6.5. What would be the sender's output (for the 2 data bits shown) if the sender's CDMA code were (1, –1, 1, –1, 1, –1, 1, –1)?

P2. Consider sender 2 in Figure 6.6. What is the sender's output to the channel (before it is added to the signal from sender 1), $Z_{i,m}^2$?

P3. Suppose that the receiver in Figure 6.6 wanted to receive the data being sent by sender 2. Show (by calculation) that the receiver is indeed able to recover sender 2's data from the aggregate channel signal by using sender 2's code.

P4. For the two-sender, two-receiver example, give an example of two CDMA codes containing 1 and −1 values that do not allow the two receivers to extract the original transmitted bits from the two CDMA senders.

P5. Suppose there are two ISPs providing WiFi access in a particular café, with each ISP operating its own AP and having its own IP address block.

a. Further suppose that by accident, each ISP has configured its AP to operate over channel 11. Will the 802.11 protocol completely break down in this situation? Discuss what happens when two stations, each associated with a different ISP, attempt to transmit at the same time.

b. Now suppose that one AP operates over channel 1 and the other over channel 11. How do your answers change?

P6. In step 4 of the CSMA/CA protocol, a station that successfully transmits a frame begins the CSMA/CA protocol for a second frame at step 2, rather than at step 1. What rationale might the designers of CSMA/CA have had in mind by having such a station not transmit the second frame immediately (if the channel is sensed idle)?

P7. Suppose an 802.11b station is configured to always reserve the channel with the RTS/CTS sequence. Suppose this station suddenly wants to transmit 1,000 bytes of data, and all other stations are idle at this time. As a function of SIFS and DIFS, and ignoring propagation delay and assuming no bit errors, calculate the time required to transmit the frame and receive the acknowledgment.

P8. Consider the scenario shown in Figure 6.33, in which there are four wireless nodes, A, B, C, and D. The radio coverage of the four nodes is shown via the shaded ovals; all nodes share the same frequency. When A transmits, it can only be heard/received by B; when B transmits, both A and C can hear/receive from B; when C transmits, both B and D can hear/receive from C; when D transmits, only C can hear/receive from D.

Suppose now that each node has an infinite supply of messages that it wants to send to each of the other nodes. If a message's destination is not an immediate neighbor, then the message must be relayed. For example, if A wants to send to D, a message from A must first be sent to B, which then sends the message to C, which then sends the message to D. Time is slotted, with a message transmission time taking exactly one time slot, e.g., as in slotted Aloha. During a slot, a node can do one of the following: (*i*) send a message; (*ii*) receive a message (if exactly one message is being sent to it), (*iii*) remain silent. As always, if a node hears two or more simultaneous transmissions, a collision occurs and none of the transmitted messages are received successfully. You can assume here that there are no bit-level errors, and thus if exactly one message is sent, it will be received correctly by those within the transmission radius of the sender.

a. Suppose now that an omniscient controller (i.e., a controller that knows the state of every node in the network) can command each node to do whatever it (the omniscient controller) wishes, i.e., to send a message, to receive a message, or to remain silent. Given this omniscient controller, what is the maximum rate at which a data message can be transferred from C to A, given that there are no other messages between any other source/destination pairs?

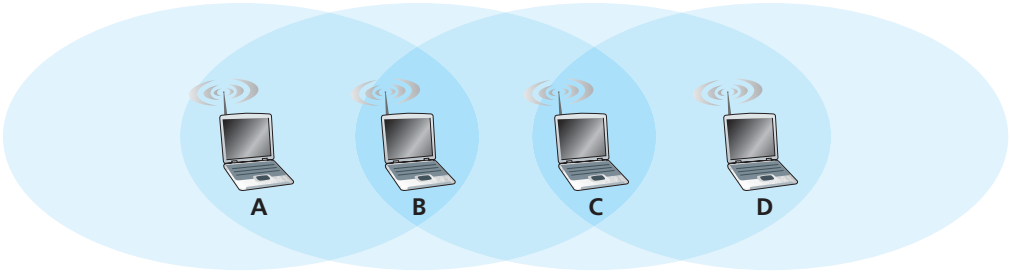


Figure 6.33 ♦ Scenario for problem P8

b. Suppose now that A sends messages to B, and D sends messages to C. What is the combined maximum rate at which data messages can flow from A to B and from D to C?

c. Suppose now that A sends messages to B, and C sends messages to D. What is the combined maximum rate at which data messages can flow from A to B and from C to D?

d. Suppose now that the wireless links are replaced by wired links. Repeat questions (a) through (c) again in this wired scenario.

e. Now suppose we are again in the wireless scenario, and that for every data message sent from source to destination, the destination will send an ACK message back to the source (e.g., as in TCP). Also suppose that each ACK message takes up one slot. Repeat questions (a) – (c) above for this scenario.

P9. Describe the format of the 802.15.1 Bluetooth frame. You will have to do some reading outside of the text to find this information. Is there anything in the frame format that inherently limits the number of active nodes in an 802.15.1 network to eight active nodes? Explain.

P10. Consider the following idealized LTE scenario. The downstream channel (see Figure 6.20) is slotted in time, across F frequencies. There are four nodes, A, B, C, and D, reachable from the base station at rates of 10 Mbps, 5 Mbps, 2.5 Mbps, and 1 Mbps, respectively, on the downstream channel. These rates assume that the base station utilizes all time slots available on all F frequencies to send to just one station. The base station has an infinite amount of data to send to each of the nodes, and can send to any one of these four nodes using any of the F frequencies during any time slot in the downstream sub-frame.

a. What is the maximum rate at which the base station can send to the nodes, assuming it can send to any node it chooses during each time slot? Is your solution fair? Explain and define what you mean by "fair."

b. If there is a fairness requirement that each node must receive an equal amount of data during each one second interval, what is the average transmission rate by the base station (to all nodes) during the downstream sub-frame? Explain how you arrived at your answer.

c. Suppose that the fairness criterion is that any node can receive at most twice as much data as any other node during the sub-frame. What is the average transmission rate by the base station (to all nodes) during the sub-frame? Explain how you arrived at your answer.

P11. In Section 6.5, one proposed solution that allowed mobile users to maintain their IP addresses as they moved among foreign networks was to have a foreign network advertise a highly specific route to the mobile user and use the existing routing infrastructure to propagate this information throughout the

network. We identified scalability as one concern. Suppose that when a mobile user moves from one network to another, the new foreign network advertises a specific route to the mobile user, and the old foreign network withdraws its route. Consider how routing information propagates in a distance-vector algorithm (particularly for the case of interdomain routing among networks that span the globe).

a. Will other routers be able to route datagrams immediately to the new foreign network as soon as the foreign network begins advertising its route?

b. Is it possible for different routers to believe that different foreign networks contain the mobile user?

c. Discuss the timescale over which other routers in the network will eventually learn the path to the mobile users.

P12. Suppose the correspondent in Figure 6.22 were mobile. Sketch the additional network-layer infrastructure that would be needed to route the datagram from the original mobile user to the (now mobile) correspondent. Show the structure of the datagram(s) between the original mobile user and the (now mobile) correspondent, as in Figure 6.23.

P13. In mobile IP, what effect will mobility have on end-to-end delays of datagrams between the source and destination?

P14. Consider the chaining example discussed at the end of Section 6.7.2. Suppose a mobile user visits foreign networks A, B, and C, and that a correspondent begins a connection to the mobile user when it is resident in foreign network A. List the sequence of messages between foreign agents, and between foreign agents and the home agent as the mobile user moves from network A to network B to network C. Next, suppose chaining is not performed, and the correspondent (as well as the home agent) must be explicitly notified of the changes in the mobile user's care-of address. List the sequence of messages that would need to be exchanged in this second scenario.

P15. Consider two mobile nodes in a foreign network having a foreign agent. Is it possible for the two mobile nodes to use the same care-of address in mobile IP? Explain your answer.

P16. In our discussion of how the VLR updated the HLR with information about the mobile's current location, what are the advantages and disadvantages of providing the MSRN as opposed to the address of the VLR to the HLR?

Wireshark Lab

At the companion Web site for this textbook, http://www.awl.com/kurose-ross, you'll find a Wireshark lab for this chapter that captures and studies the 802.11 frames exchanged between a wireless laptop and an access point.

AN INTERVIEW WITH...

Deborah Estrin

Deborah Estrin is Professor of Computer Science at UCLA, the Jon Postel Chair in Computer Networks, Director of the Center for Embedded Networked Sensing (CENS), and co-founder of the non-profit openmhealth.org. She received her Ph.D. (1985) in Computer Science from M.I.T., and her B.S. (1980) from UC Berkeley. Estrin's early research focused on the design of network protocols, including multicast and inter-domain routing. In 2002 Estrin founded the NSF-funded Science and Technology Center, CENS (http://cens.ucla.edu), to develop and explore environmental monitoring technologies and applications. Currently Estrin and collaborators are developing participatory sensing systems, leveraging the programmability, proximity, and pervasiveness of mobile phones; the primary deployment contexts are mobile health (http://openmhealth.org), community data gathering, and STEM education (http://mobilizingcs.org). Professor Estrin is an elected member of the American Academy of Arts and Sciences (2007) and the National Academy of Engineering (2009). She is a fellow of the IEEE, ACM, and AAAS. She was selected as the first ACM-W Athena Lecturer (2006), awarded the Anita Borg Institute's Women of Vision Award for Innovation (2007), inducted into the WITI hall of fame (2008) and awarded Doctor Honoris Causa from EPFL (2008) and Uppsala University (2011).

Please describe a few of the most exciting projects you have worked on during your career. What were the biggest challenges?

In the mid-90s at USC and ISI, I had the great fortune to work with the likes of Steve Deering, Mark Handley, and Van Jacobson on the design of multicast routing protocols (in particular, PIM). I tried to carry many of the architectural design lessons from multicast into the design of ecological monitoring arrays, where for the first time I really began to take applications and multidisciplinary research seriously. That interest in jointly innovating in the social and technological space is what interests me so much about my latest area of research, mobile health. The challenges in these projects were as diverse as the problem domains, but what they all had in common was the need to keep our eyes open to whether we had the problem definition right as we iterated between design and deployment, prototype and pilot. None of them were problems that could be solved analytically, with simulation or even in constructed laboratory experiments. They all challenged our ability to retain

clean architectures in the presence of messy problems and contexts, and they all called for extensive collaboration.

What changes and innovations do you see happening in wireless networks and mobility in the future?

I have never put much faith into predicting the future, but I would say we might see the end of feature phones (i.e., those that are not programmable and are used only for voice and text messaging) as smart phones become more and more powerful and the primary point of Internet access for many. I also think that we will see the continued proliferation of embedded SIMs by which all sorts of devices have the ability to communicate via the cellular network at low data rates.

Where do you see the future of networking and the Internet?

The efforts in named data and software-defined networking will emerge to create a more manageable, evolvable, and richer infrastructure and more generally represent moving the role of architecture higher up in the stack. In the beginnings of the Internet, architecture was layer 4 and below, with applications being more siloed/monolithic, sitting on top. Now data and analytics dominate transport.

What people inspired you professionally?

There are three people who come to mind. First, Dave Clark, the secret sauce and unsung hero of the Internet community. I was lucky to be around in the early days to see him act as the "organizing principle" of the IAB and Internet governance; the priest of rough consensus and running code. Second, Scott Shenker, for his intellectual brilliance, integrity, and persistence. I strive for, but rarely attain, his clarity in defining problems and solutions. He is always the first person I email for advice on matters large and small. Third, my sister Judy Estrin, who had the creativity and courage to spend her career bringing ideas and concepts to market. Without the Judys of the world the Internet technologies would never have transformed our lives.

What are your recommendations for students who want careers in computer science and networking?

First, build a strong foundation in your academic work, balanced with any and every real-world work experience you can get. As you look for a working environment, seek opportunities in problem areas you really care about and with smart teams that you can learn from.

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