CHAPTER 4

Wireless Medium Access Alternatives

4.1 INTRODUCTION

This chapter presents an overview of the access methods commonly used in wireless networks. Access methods form a part of Layer 2 of the OSI protocol stack and Layer 3 of the IEEE 802 standard for LANs that is responsible for interacting with the medium to coordinate the successful operation of multiple terminals over the wireless channel. Most multiple access methods were originally developed for wired networks and later on adopted to the wireless medium. However, requirements on the wired and wireless media are different, thereby demanding modifications in the original protocols to make them suitable for the wireless medium. Today the main differences between wireless and wired channels are availability of bandwidth and reliability of transmissions. The wired medium is moving toward optical media with enormous bandwidth and very reliable transmission. Bandwidth in wireless systems is always limited because the medium (air) cannot be duplicated, and the medium is shared between all wireless systems, including multichannel broadcast television and a number of other bandwidth demanding applications and services. In the case of wired operation, we can always lay additional cable to increase the capacity as needed even if it is an expensive proposition. In a wireless environment, we can reduce the size of *cells* to increase capacity as discussed in Chapter 5. With the reduction of the size of the cells, the number of cells increases, and the need for improvements in the wired infrastructure to connect these cells increases. Also the complexity of the network for handling additional handoffs and mobility management increases posing a practical limitation upon the maximum capacity of the network. As far as transmission reliability is concerned, as we saw in Chapter 2, the wireless medium always suffers from multipath and fading, which causes a serious threat to reliable data transmission over the communication link. Because the wireless channel is so unreliable, as discussed in Chapter 3, people have developed a number of signal processing techniques to improve transmission reliability over the wireless channel. In spite of these techniques, the reliability of the wireless medium is far below that of the wired medium used as the backbone of the wireless networks.

Although in practice we prefer to have the same access method and the same frame structure for wired backbone and the wireless access, wireless networks often use different packet sizes and a modified access method to optimize the performance to the specifics of the unreliable wireless medium.

Example 4.1: IEEE 802.3 and IEEE 802.11

The IEEE 802.3 standard (based on Ethernet) is the successful and dominant standard for wired LANs. Consequently, the IEEE 802.11 WLAN standard, in ideal situations, desired using the same access method as previously established with IEEE 802.3. Carrier sense multiple-access with collision detection (CSMA/CD) is the protocol used in the Ethernet. However, collision detection in wireless channels is extremely challenging, and IEEE 802.11 had to resort to carrier sense multiple-access with collision avoidance (CSMA/CA) that can be viewed as a wireless adaptation of IEEE 802.3.

Example 4.2: ATM and Wireless ATM

In the 1990s, ATM was perceived to be the transmission scheme for all future networking. In the mid 1990s when wireless solutions were considered, a wireless ATM working group was formed to extend the ATM short packet solution with QoS for wireless access. The group had to make significant compromises as discussed in Chapter 12 because ATM was designed for broadband and reliable transmission over optical channels.

To avoid substantial overlap with existing literature, we describe access methods used in wireless networks with justification of why and how they are employed in different wireless networks.

As we explained in previous chapters, wireless networks have evolved around voice or data applications, and as a result we can divide them into voice- and data-oriented networks. The access methods adopted by voice- and data-oriented networks are quite different. Voice-oriented networks are designed for relatively long telephone conversations as the main application. Each communication session for this application exchanges several megabytes of information in both directions. A signaling channel that exchanges short messages between two calling components sets up the call by obtaining resources (such as the link, switches, etc.) in the telephone network at the beginning of the conversation and terminates these arrangements by releasing the resources at the end of the call. The wireless access methods evolved for interaction with these networks assigns a slot of time, a portion of frequency, or a specific code to a user preferably for the entire length of the conversation. We refer to these techniques as fixed-assignment channel access methods or channel partitioning techniques. Data networks were originally designed for bursts of data for which the supporting network does not have a separate signaling channel. In packet communications each packet carries some "signaling information" related to the address of the destination and the source. We refer to the access methods used in these networks as random-access methods accommodating randomly arriving packets of data. Certain local area data networks also *take turns* in accessing the medium as in the case of token passing and polling schemes. In some other cases, the random access mechanisms are used to temporarily *reserve* the medium for transmitting the packet. In the next two sections of this chapter, we provide a short description of the fixed-assignment and random access methods used in voice- and data-oriented wireless networks, respectively.

4.2 FIXED-ASSIGNMENT ACCESS FOR VOICE-ORIENTED NETWORKS

All existing voice-oriented wireless networks such as cellular telephony or PCS services use fixed-assignment channel access or channel partitioning techniques. In the fixed-assignment access method, a fixed allocation of channel resources, frequency, time, or a spread spectrum code are made available on a predetermined basis to a single user for the duration of the communication session. The three basic fixed-assignment multiple-access methods are FDMA, TDMA, and CDMA. The choice of an access method will have a great impact on the capacity and QoS provided by a network. The impact of multiple access schemes is so important that we commonly refer to various voice-oriented wireless systems by their channel access method, which is only a part of the layer two specification of the air interface of the network.

Example 4.3: Common Terminology for Digital Cellular Systems

The GSM and the North American IS-136 digital cellular standards are commonly referred to as digital TDMA cellular systems and the IS-95/IMT-2000 standards are called digital CDMA cellular systems.

In reality, different systems use different modulation techniques as well. However, as we will see in the rest of this book, the impact of the choice of access method on the capacity and overall performance of the network is much more profound. Consequently, the system is really distinguished by its access method. As we will see in our examples of cellular networks, a network that is identified with an access technique often uses other random or fixed assignment techniques as a part of its overall operation. However, it is identified by the access techniques employed for transferring the main information source for which the network is designed to carry.

Example 4.4: Random Access Techniques in Cellular Networks

GSM uses slotted ALOHA (a random access method) to establish a link between the mobile terminal and the base station. It also has an optional frequency-hopping pattern that improves the system performance when there is fading of the radio signal. However, the GSM network is built for voice communications, and each session uses TDMA as the access method.

Another important design parameter related to the access method is the differentiation between the carrier frequencies of the forward (downlink—communication between the base station and mobile terminals) and reverse (uplink—communication between the mobile terminal and the base station) channels. If both forward and reverse channels use the same frequency band for communications, but the forward and reverse channels employ alternating time slots, the system is referred to as employing TDD. If the forward and reverse channels use different carrier frequencies that are sufficiently separated, the duplexing scheme is referred to as FDD. With TDD, because only one frequency carrier is needed for a duplex operation, we can share more of the RF circuitry between the forward and the reverse channels. The reciprocity of the channel in TDD allows for exact open-loop power control and simultaneous synchronization of the forward and reverse channels. TDD techniques are used in systems intended for low-power local area communications where interference must be carefully controlled and where low complexity and low-power consumption are very important. Thus TDD systems are often used in local area pico- or microcellular systems deployed by PCS networks. FDD is mostly used in macrocellular systems designed for coverage of several tens of kilometers where implementation of TDD is more challenging (see Fig. 4.1).

4.2.1 Frequency Division Multiple Access (FDMA)

In an FDMA environment, all users can transmit signals simultaneously, and they are separated from one another by their frequency of operation. The FDMA technique is built upon FDM. FDM is the oldest and still a commonly used multiplexing technique in the trunks connecting switches in the PSTN. It is also the choice of radio and TV broadcast, as well as cable TV distribution. FDM is more suitable for analog technology because it is easier to implement. When FDM is used for channel access, it is referred to as FDMA.

Example 4.5: FDMA in AMPS with FDD

Figure 4.1(a) shows the FDMA/FDD system commonly used in 1G analog cellular systems such as AMPS and a number of early cordless telephones. In FDMA/FDD systems, forward and reverse channels use different carrier frequencies, and a fixed subchannel pair is assigned to a user terminal during the communication session. At the receiving end, the mobile terminal filters the designated channel out of the composite signal. As shown in Figure 4.2(a), the AMPS system allocates 30 kHz of bandwidth for each forward and reverse channel. There are a total of 421 channels in 25 MHz of spectrum assigned to each direction; 395 of these channels are used for the voice traffic and the rest for signaling.

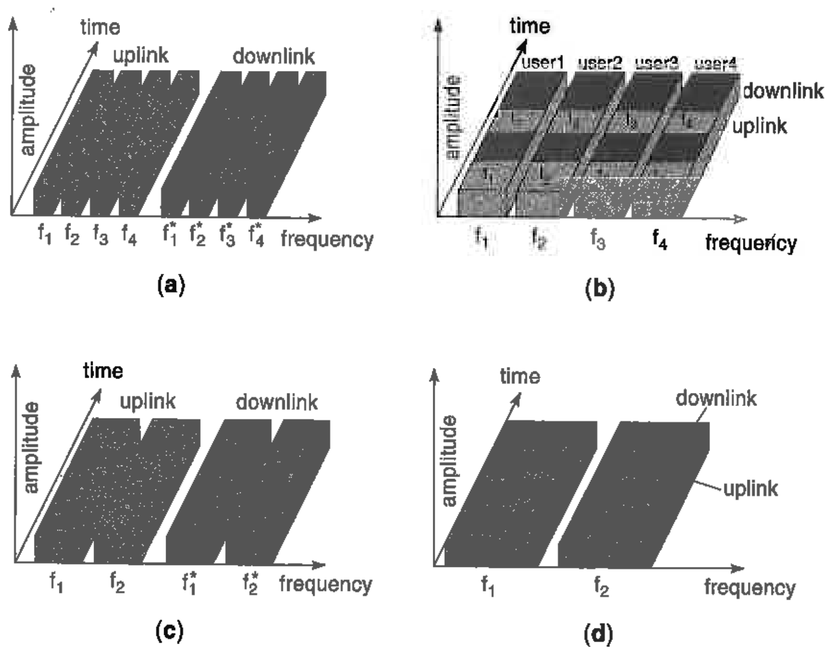

Figure 4.1 (a) FDMA/FDD, (b) FDMA/TDD, (c) TDMA/FDD with multiple carriers, (d) TDMA/TDD with multiple carriers.

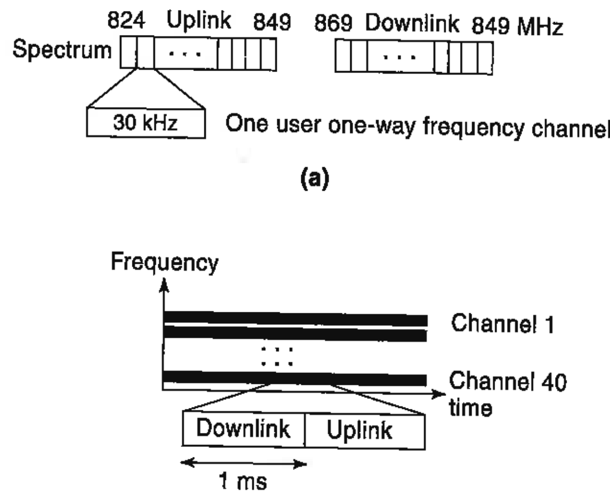

Figure 4.2 (a) FDMA/FDD in AMPS and (b) FDMA/TDD in CT-2.

Example 4.6: FDMA in CT-2 with TDD

Figure 4.1(b) shows an FDMA/TDD system used in the CT-2 digital cordless telephony standard. Each user employs a single carrier frequency for all communications. The forward and reverse transmissions take turns via alternating time slots. This system was designed for distances of up to 100 meters, and a voice conversation is based on 32 kbps ADPCM voice coding. As shown in Figure 4.2(b) the total allocated bandwidth for CT-2 is 4 MHz, supporting 40 carriers each using 100 kHz of bandwidth.

The designer of an FDMA system must pay special attention to adjacent channel interference, in particular in the reverse channel. In both forward and reverse channels, the signal transmitted must be kept confined within its assigned band, at least to the extent that the out-of-band energy causes negligible interference to the users employing adjacent channels. Operation of the forward channel in wireless FDMA networks is very similar to wired FDM networks. In forward wireless channels, in a manner similar to that of wired FDM systems, the signal received by all mobile terminals has the same received power, and interference is controlled by adjusting the sharpness of the transmitter and receiver filters for the separate carrier frequencies. The problem of adjacent channel interference is much more challenging on the reverse channel. On the reverse channel, mobile terminals will be operating at different distances from the BS. The RSS at the BS, of a signal transmitted by a mobile terminal close to the BS, and the RSS at the BS of a transmission by a mobile terminal at edges of the cell are often substantially different, causing problems in detecting the weaker signal. This problem is usually referred to as the near-far problem. If the out-of-band emissions are large, they may swamp the actual information-carrying signal.

Problem 1: Near-Far Problem

a) What is the difference between the received signal strength of two terminals located in 10 m and 1 km from a base station in an open area?

b) Explain the effects of shadow fading on the difference in the RSS.

c) What would be the impact if the two terminals were operating in two adjacent channels? Assume out-of-band radiation that is 40 dB below the main lobe.

Solution:

a) As we saw in Chapter 2, the received signal strength falls by around 40 dB per decade of distance in open areas. Therefore, the received powers from a mobile terminal that is 10 meters from a BS and another, that is at a distance of 1 km, are 80 dB apart.

b) In addition to the fall of the RSS with distance, we also discussed the issues of multipath and shadow fading in radio channels that cause power fluctuations on the order of several tens of dBs. Therefore, the difference in the received powers due to the near-far problem may exceed even 100 dB.

c) If the out-of-band emission is only 40 dB below that of the transmitted power, it may exceed the strength of the information-bearing signal by almost 60 dB.

To handle the near-far problem, FDMA cellular systems adopt two different measures. First, when frequencies are assigned to a cell, they are grouped such that the frequencies in each cell are as far apart as possible. The second measure employed is power control that is discussed in Chapter 6. In addition, whenever FDMA is employed, *guard bands* are included in the frequency channel to further reduce adjacent channel interference. This, however, has the effect of reducing the overall spectrum efficiency.

4.2.2 Time Division Multiple Access (TDMA)

In TDMA systems, a number of users share the same frequency band by taking assigned turns in using the channel. The TDMA technique is built upon the TDM scheme commonly used in the trunks for the telephones systems. The major advantage of the TDMA over FDMA is its format flexibility. Because of the fully digital format and the flexibility of buffering and multiplexing functions, time-slot assignments among multiple users are readily adjustable to provide different access rates for different users. This feature is particularly adopted in the PSTN, and the TDM scheme forms the backbone of all digital connections in the heart of the PSTN. The hierarchy of digital transmission trunks used in North America is the so-called T-carrier system that has an equivalent European system (the E-carriers) approved by the ITU. In the hierarchy of digital transmission rates standardized throughout North America, the basic building block is the 1.544 Mbps link known as T-1 carrier. A T-1 transmission frame is formed by TDD 24 PCM-encoded voice channels, each carrying 64 kbps of users data. Service providers often lease T-carriers to interconnect their own switches and routers and for forming their own networks.

Example 4.7: The Use of T-carriers in Cellular Networks

Cellular networks often lease T-carriers from the long-haul telephone companies to interconnect their own switches referred to as mobile switching centers

(MSCs). The difference between the MSC and a regular switch in the PSTN is that the MSC can support mobility of the terminal. The details of these differences are discussed in later chapters when we provide examples of cellular networks. The end-user subscribes to the cellular service provider.

Example 4.8: The Use of T-1 Lines in the Internet

The routers in the Internet are sometimes connected through leased T-carrier telephone lines to form part of the Internet. The difference between a router and a PSTN switch is that the router can handle packet switching whereas the PSTN switch uses circuit switching. The end-user subscribes to an Internet service provider (ISP) in this case.

With TDMA, a transmit controller assigns time slots to users, and an assigned time slot is held by a user until the user releases it. At the receiving end, a receiver station synchronizes to the TDMA signal frame, and extracts the time slot designated for that user. The heart of this operation is synchronization that was not needed for FDMA systems. The TDMA concept was developed in the 1960s for use in digital satellite communication systems and first became operational commercially in telephone networks in the mid-1970s [PAH95].

In cellular and cordless systems, the migration to TDMA from FDMA took place in the 2G systems. The first cellular standard adopting TDMA was GSM. The GSM standard was initiated to support international roaming among Scandinavian countries in particular and the rest of Europe in general. The digital voice adoption in TDMA format facilitated the network implementation, resulted in improvements in the quality of the voice, and provided a flexible format to integrate data services in the cellular network. The FDMA systems in the United States very quickly observed a capacity crunch in major cities, and among the options for increasing capacity, TDMA was adopted initially through the IS-54 system that was later on replaced by IS-136. TDMA was adopted in 2G cordless telephones such as DECT to provide format flexibility and to allow more compact and low-power terminals.

Example 4.9: TDMA in GSM

Figure 4.1(c) shows an FDMA/TDMA/FDD channel used in 2G digital cellular networks. Figure 4.3 shows a particular example of the 8-slot TDMA scheme used in the GSM system. Forward and reverse channels use separate carrier frequencies (FDD). Each carrier can support up to eight simultaneous users via TDMA, each using a 13 kbps encoded digital speech, within a 200 MHz carrier bandwidth. A total of 124 frequency carriers (FDMA) are available in the 25 MHz allocated band in each direction. One hundred kHz of band is allocated as a guard band at each edge of the overall allocated band.

Example 4.10: TDMA in DECT

Figure 4.1(d) shows an FDMA/TDMA/TDD system used in the Pan-European digital PCS standard DECT. Because distances are short, a TDD format allows

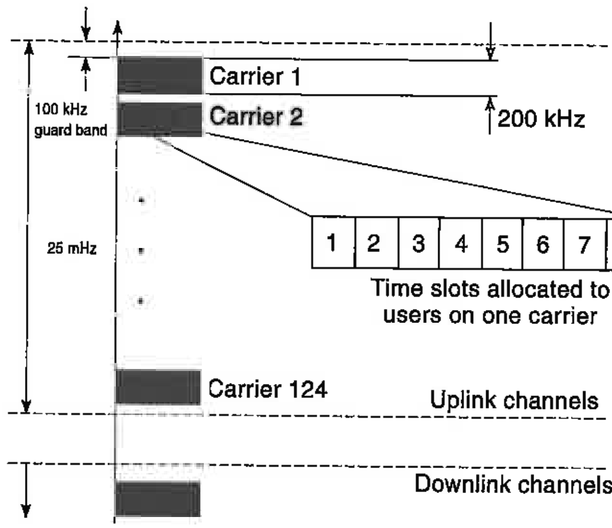

Figure 4.3 FDMA/TDMA/FDD in GSM.

using the same frequency for forward and reverse operations. The bandwidth per carrier is 1.728 MHz which can support up to 12 ADPCM coded speech channels via TDMA. The total allocated band in the EC is 10 MHz that can support five carriers (FDMA). Figure 4.4 shows the details of the TDMA/TDD time slots use in the DECT system. The frame duration is 10 ms, with 5 ms for portable-to-fixed station and 5 ms for fixed-to-portable. The transmitter transfers information in signal bursts which it transmits in slots of duration 10/24 = 0.417 ms. With 480 bits per slot (including a 64-bit guard time), the total bit rate is 1.152 Mbps. Each slot contains 64 bits for system control (C, P, Q, and M channels) and 320 bits for user information (I channel).

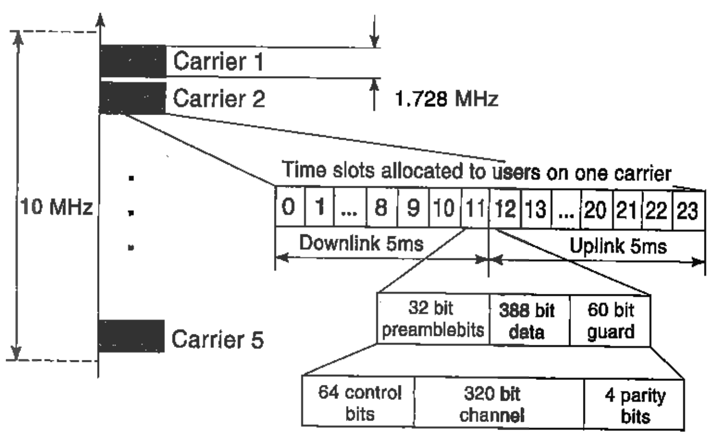

Figure 4.4 FDMA/TDMA/TDD in DECT.

Example 4.11: TDMA in IS-136

Figure 4.5 shows the frame format for the TDMA/FDD with six slots considered for IS-136 both for the forward (base to mobile) and reverse (mobile to base) channels. In IS-136 each 30 kHz digital channel has a channel transmission rate of 48.6 kbps. The 48.6 kbps stream is divided into six TDMA channels of 8.1 kbps each. The IS-136 slot and frame format, shown in Figure 4.5, is much simpler than that of the GSM standard. The 40-ms frame is composed of six 6.67-ms time slots. Each slot contains 324 bits, including 260 bits of user data, and 12 bits of system control information in a slow associated control channel (SACCH). There is also a 28-bit synchronization sequence, and a 12-bit digital verification color code (DVCC) used to identify the frequency channel to which the mobile terminal is tuned. In the mobile-to-base direction, the slot also contains a guard time interval of a six-bit duration when no signal is transmitted, and a six-bit ramp interval to allow the transmitter to reach its full output power level.

Due to the near-far problem, the received signal on the reverse channel from a user occupying a time slot can be much larger than the received power from the terminal using the adjacent time slot. In such a case, the receiver will have difficulties in distinguishing the weaker signal from the background noise. In a manner similar to FDMA systems, TDMA systems also use power control to handle this near-far problem.

4.2.3 Code-Division Multiple Access (CDMA)

With the growing interest in the integration of voice, data, and video traffic in telecommunication networks, CDMA appears increasingly attractive as the wireless access method of choice. Fundamentally, integration of various types of traffic is readily accomplished in a CDMA environment as coexistence in such an

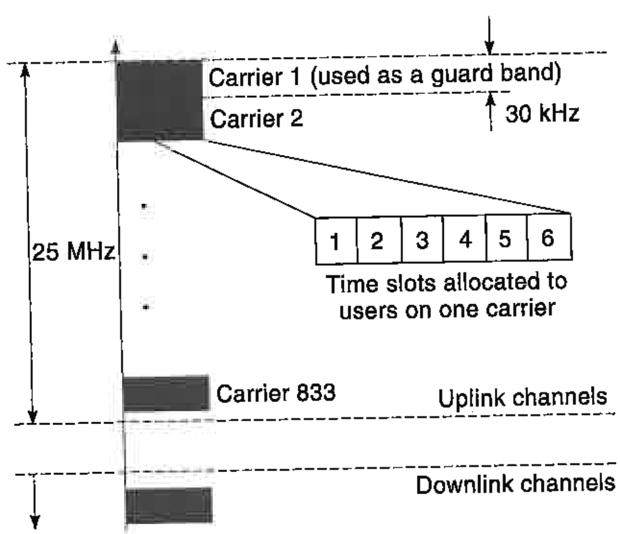

Figure 4.5 FDMA/TDMA/FDD in IS-136 standard.

environment does not require any specific coordination among user terminals. In principle, CDMA can accommodate various wireless users with different bandwidth requirements, switching methods and technical characteristics without any need for coordination. Of course, because each user signal contributes to the interference seen by other users, power control techniques are essential in the efficient operation of a CDMA system.

To illustrate CDMA and how it is related to FDMA and TDMA, it is useful to think of the available band and time as resources we use to share among multiple users. In FDMA, the frequency band is divided into slots, and each user occupies that frequency throughout the communication session. In TDMA, a larger frequency band is shared among the terminals, and each user uses a slot of time during the communication session. As shown in Figure 4.6, in a CDMA environment multiple users use the same band at the same time, and the user is differentiated by a code that acts as the key to identify that user. These codes are selected so that when they are used at the same time in the same band a receiver knowing the code of a particular user can detect that user among all the received signals. In the CDMA/FDD [Figure 4.7(a)] that is used in IS-95 and IMT-2000, the forward and reverse channels use different carrier frequencies. If both transmitter and receiver use the same carrier frequency [Figure 4.7(b)], the system is CDMA/TDD.

In CDMA, each user is a source of noise to the receiver of other users, and if we increase the number of users beyond a certain value, the entire system collapses because the signal received in each specific receiver will be buried under the noise caused by many other users. An important question is, how many users can simultaneously use a CDMA system before the system collapses?

4.2.3.1 Capacity of CDMA

CDMA systems are implemented based on the spread spectrum technology that is presented in Chapter 3. In its most simplified form, a spread spectrum transmitter spreads the signal power over a spectrum N times wider than the spectrum

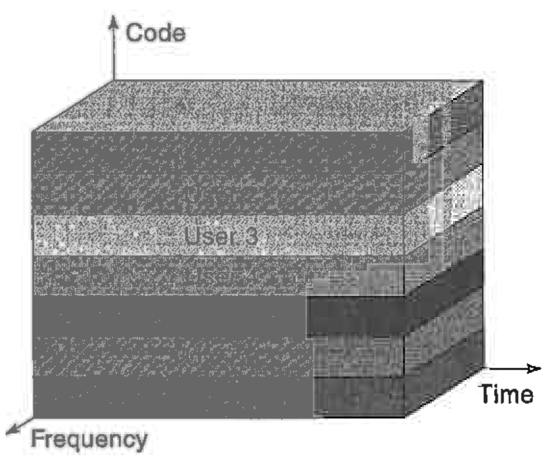

Figure 4.6 Simple illustration of CDMA.

of the message signal. In other words, an information bandwidth of R occupies a transmission bandwidth of W, where:

$$W = NR \tag{4.1}$$

The spread spectrum receiver processes the received signal with a *processing gain* of N. This means that during the processing at the receiver, the power of the received signal having the code of that particular receiver will be increased N times beyond the value before processing.

Let us consider the situation of a single cell in a cellular system employing CDMA. Assume that we have M simultaneous users on the reverse channel of a CDMA network. Further let us assume that we have an ideal power control enforced on the channel so that the received power of signals from all terminals has the same value P. Then, the received power from the target user after processing at the receiver is NP, and the received interference from $M - 1$ other terminals is $(M - 1)P$. If we also assume that a cellular system is interference limited and the background noise is dominated by the interference noise from other users, the received signal-to-interference ratio for the target receiver will be:

$$S_r = \frac{NP}{(M-1)P} = \frac{N}{M-1} \tag{4.2}$$

All users always have a requirement for the acceptable error rate of the received data stream. For a given modulation and coding specification of the system, that error rate requirement will be supported by a minimum S_r requirement that can be used in Eq. (4.2) to solve for the number of simultaneous users. Then, solving Eqs. (4.1) and (4.2) for M, we will have:

$$M = \frac{W}{R}\frac{1}{S_r} + 1 \cong \frac{W}{R}\frac{1}{S_r} \tag{4.3}$$

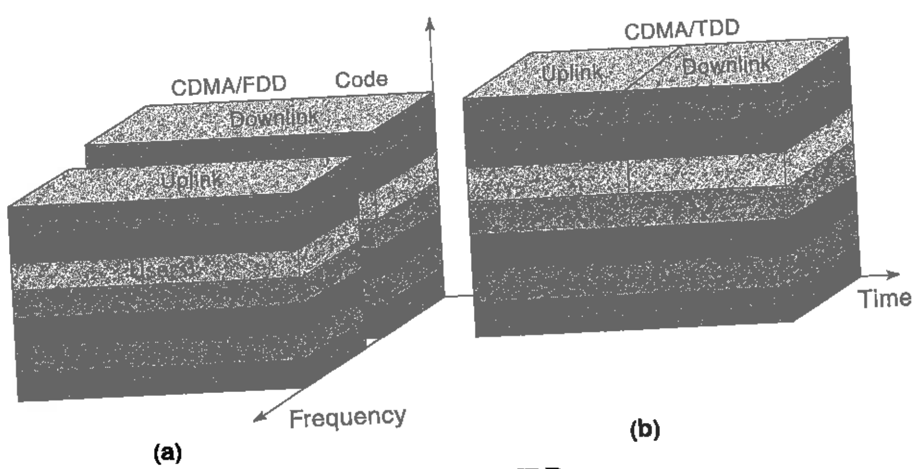

Figure 4.7 (a) CDMA/FDD and (b) CDMA/TDD.

Problem 2: Capacity of One Carrier in a Single-Cell CDMA System

Using QPSK modulation and convolutional coding, the IS-95 digital cellular systems require 3 dB < S_r < 9 dB. The bandwidth of the channel is 1.25 MHz, and the transmission rate is R = 9600 bps. Find the capacity of a single IS-95 cell.

Solution:

Using Equation (4.3) we can support from up to

$$M = \frac{1.25\text{ MHz}}{9600\text{ bps}}\frac{1}{8} \approx 16 \text{ to } M = \frac{1.25\text{ MHz}}{9600\text{ bps}}\frac{1}{2} \approx 65 \text{ users.}$$

4.2.3.2 Practical Considerations

In the practical design of digital cellular systems, three other parameters affect the number of users that can be supported by the system, as well as the bandwidth efficiency of the system. These are the number of sectors in each base station antenna, the voice activity factor, and the interference increase factor. These parameters are quantified as factors used in the calculation of the number of simultaneous users that the CDMA system can support. The use of sectored antennas is an important factor in maximizing bandwidth efficiency. Cell sectorization using directional antennas reduces the overall interference, increasing the allowable number of simultaneous users by a *sectorization gain factor,* which we denote by G_A. With ideal sectorization the users in one sector of a base station antenna do not interfere with the users operating in other sectors, and $G_A = N_{sec}$ where N_{sec} is the number of sectors in the cell. In practice antenna patterns cannot be designed to have ideal characteristics, and due to multipath reflections, users in general communicate with more than one sector. Three-sector base station antennas are commonly used in cellular systems, and a typical value of the sectorization gain factor is assumed to be $G_A = 2.5$ (4 dB). The voice activity interference reduction factor G_v is the ratio of the total connection time to the active talkspurt time. On the average, in a two-way conversation, each user talks roughly 50 percent of the time. The short pauses in the flow of natural speech reduce the activity factor further to about 40 percent of the connection time in each direction. As a result, the typical number used for G_v is 2.5 (4 dB). The interference increase factor H_0 accounts for users in other cells in the CDMA system. Because all neighboring cells in a CDMA cellular network operate at the same frequency, they will cause additional interference. This interference is relatively small due to the processing gain of the system and the distances involved; a value of $H_0 = 1.6$ (2 dB) is commonly used in the industry.

Incorporating these three factors as a correction to Equation (4.3), the number of simultaneous users that can be supported in a CDMA cell can be approximated by

$$M = \frac{W}{R}\frac{1}{S_r} + 1 \cong \frac{W}{R}\frac{1}{S_r}\frac{G_A G_v}{H_0} \tag{4.4}$$

If we define the *performance improvement factor* in a digital cellular system as

$$K = \frac{G_A G_v}{H_0} \tag{4.5}$$

assuming the typical parameter values given earlier, the performance improvement factor is $K = 4$ (6 dB).

Problem 3: Capacity of One Carrier in a Multi-Cell CDMA System with Correction Factors

Determine the multicell IS-95 CDMA capacity with correction for sectorization and voice activity. Use the numbers from Problem 2.

Solution:

If we continue the previous example with the new correction factor included, the range for the number of simultaneous users becomes $64 < M < 260$.

4.2.4 Comparison of CDMA, TDMA, and FDMA

With the success of IS-95 CDMA systems in its challenge to conventional IS-136 TDMA systems in the United States and the adoption of W-CDMA as the primary choice for the 3G cellular networks, one wonders why CDMA has become the favorite choice for wireless access in voice-oriented networks. Spread spectrum technology became the favorite technology for military applications because of its capability to provide a low probability of interception and strong resistance to interference from jamming. In the cellular industry, CDMA was introduced as an alternative to TDMA to improve the capacity of 2G cellular systems in the United States. As a result, much of the early debates in this area were focused on calculation of the capacity of CDMA as it is compared with TDMA. However, capacity is not the only reason for the success of the CDMA technology. As a matter of fact, calculation of the capacity of CDMA using the simple approach provided earlier is *not* very conclusive and is subject to a number of assumptions such as perfect power control that cannot be practically met. The first CDMA service providers in the United States were using slogans such as "you cannot believe your ears!" to address the superior quality of voice for the CDMA. However, the superiority of voice is partially dependent on the speech coder, and it is not a CDMA versus TDMA issue. In order to provide a good explanation for the success of a complex and multidisciplinary technology, such as a cellular network, addressing consumer market issues has always been very important. Those of us involved in this debate for the past decade have seen the discussion of the ups and downs of CDMA in variety of forums. One of the most interesting events that the principal author remembers was in 1997 in a major wireless conference in Taipei where one the most famous figures in this debate in his keynote speech at the opening of the conference declared that "we have seen in the past that the VHS which was not a better technology defeated BETA." In his perception, at that time, CDMA was similar to BETA. In less than a year or so after that, CDMA was selected by a number of different communities around the world as the technology of choice for 3G and IMT-2000.

In the rest of this section, we bring out a number of issues that may enlighten the reader toward a deeper understanding of the technical aspects of CDMA systems as they are compared with TDMA and FDMA networks. We hope that this may lead the reader to her/his own conclusion about the success of CDMA.

Format Flexibility. Telephone voice was the dominant source of income for the telecommunication industry up to the end of the past century. In the new millennium, the strong emergence of Internet and cable TV industries has created a case for other popular multimedia applications. The cellular phones that were designed for telephony applications are now being used for other applications and need support for multimedia applications. To support a variety of data rates with different requirements, a network needs format flexibility. As we discussed earlier, one of the reasons for migrating from analog FDMA to digital TDMA was that TDMA provides a more flexible environment for integration of voice and data. The time slots of a TDMA network designed for voice transmission can be used individually or in a group format to transmit data from users and to support different data rates. However, all these users should be time synchronized and the quality of the transmission channel is the same for all of them. The chief advantage of CDMA relative to TDMA is its flexibility in timing and the quality of transmission. In CDMA users are separated by their codes, unaffected by the transmission time relative to other users. The power of the user can also be adjusted with respect to others to support a certain quality of transmission. In CDMA each user is far more liberated from the other users, allowing a fertile setting to accommodate different service requirements to support a variety of transmission rates with different qualities of transmission to support multimedia or any other emerging application.

Performance in Multipath Fading. As we saw in Chapter 2, multipath in wireless channels causes frequency selective fading. In frequency selective fading, when the transmission band of a narrowband system coincides with the location of the fade, no useful signal is received. As we increase the transmission bandwidth, fading will occupy only a portion of the transmission band, providing an opportunity for a wideband receiver to take advantage of the portion of the transmission band not under fade and a more reliable communication link. In Chapter 3 we introduced DFE, OFDM, sectored antennas, and spread spectrum as technologies that can be employed in wideband systems to handle frequency selective fading. The wider the bandwidth, the better is the opportunity for averaging out the faded frequency.

These technologies are not used in the 1G analog cellular FDMA systems because they were analog systems and these techniques are digital. The Pan European GSM digital cellular system uses 200 kHz of band, and the standard recommends using DFE. The North American digital cellular system, IS-136, uses digital transmission over the same analog band of 30 kHz of the North American AMPS system and does not recommend equalization because the bandwidth is not very large. An equalizer needs additional circuitry, and some power budget at the receiver that was one of the drawbacks considered in IS-136. The bandwidth of the IS-95 CDMA system is 1.25MHz and W-CDMA systems for 3G networks use bandwidths that are as high 10 MHz. RAKE receivers are used to increase the benefits of wideband transmission by taking advantage of the so-called in-band or time diversity of the wideband signal. This is one of the reasons for having a better quality of voice in CDMA systems. As we mentioned earlier, quality of voice is also effected by the robustness of the speech-coding algorithm, coverage of service, methods to handle interference, handoffs, and power control as well.

System Capacity. Comparison of the capacity depends on a number of issues, including the frequency reuse factor, speech coding rate, and the type of antenna. Therefore a fair comparison would be difficult unless we go to practical systems. The following simple example compares the capacity of FDMA (AMPS), TDMA (IS-136), and CDMA (IS-95) used in debates to evaluate alternatives for the 2G North American digital cellular systems to replace the 1G analog.

Problem 4: Comparison of the Capacity of Different 2G Systems

Compare the capacity of IS-95 CDMA with AMPS FDMA and IS-136 TDMA systems. For the CDMA system, assume an acceptable signal to interference ratio of 6 dB, data rate of 9600 bps, voice duty cycle of 50 percent, effective antenna separation factor of 2.75 (close to ideal 3-sector antenna), and neighboring cell interference factor of 1.67.

Solution:

For the IS-95 CDMA using Equation (4.4) for each carrier with $W = 1.25$ MHz, $R = 9600$ bps, $S_r = 4$ (6dB), $G_v = 2$ (50 percent voice activity), $G_A = 2.75$, and $H_0 = 1.67$ we have:

$$M = \frac{W}{R}\frac{1}{S_r}\frac{G_A G_v}{H_0} = 108 \text{ users per cell}$$

For the IS-136 with a carrier bandwidth of $W_c = 30$ kHz, the number of users per carrier of $N_u = 3$, and frequency reuse factor of $K = 4$ (commonly used in these systems), each $W = 1.25$ MHz of bandwidth provides for

$$M = \frac{W}{W_c}\frac{N_u}{K} = 31.25 \text{ users per cell}$$

For the AMPS analog system with carrier bandwidth of $W_c = 30$ kHz, and frequency reuse factor of $K = 7$ (commonly used in these systems), each $W = 1.25$ MHz of bandwidth provides for

$$M = \frac{W}{W_c}\frac{1}{K} = 6 \text{ users per channel}$$

Another example of this form is instructive to compare these systems with the GSM.

Problem 5: Comparison of NA Systems with GSM

Determine the capacity of GSM for $K = 3$.

Solution:

For the GSM system with a carrier bandwidth of $W_c = 200$ kHz, the number of users per carrier of $N_u = 8$, and frequency reuse factor of $K = 3$ (commonly used in these systems), each $W = 1.25$MHz of bandwidth provides for

$$M = \frac{W}{W_c}\frac{N_u}{K} = 16.7 \text{ users per cell}$$

Handoff. As we discuss in Chapter 6, handoff occurs when a received signal in an MS becomes weak and another BS can provide a stronger signal to the MS. The 1G FDMA cellular systems often used the so-called hard-decision handoff in which the base station controller monitors the received signal from the BS and at the appropriate time switches the connection from one BS to another. TDMA systems use the so-called *mobile-assisted handoff* in which the mobile station monitors the received signal from available BSs and reports it to the base station controller which then makes a decision on the handoff. Because adjacent cells in both FDMA and TDMA use different frequencies, the MS has to disconnect from and reconnect to the network that will appear as a click to the user. Handoffs occur at the edge of the cells when the received signals from both BSs are weak. The signals also fluctuate anyway because they are arriving over radio channels. As a result, decision making for the handoff time is often complex, and the user experiences a period of poor signal quality and possibly several clicks during the completion of the handoff process. Because adjacent cells in a CDMA network use the same frequency, a mobile moving from one cell to another can make "seamless" handoff by the use of signal combining. When the mobile station approaches the boundary between cells, it communicates with both cells. A controller combines the signals from both links to form a better communication link. When a reliable link has been established with the new base station, the mobile stops communicating with the previous base station, and communication is fully established with the new base station. This technique is referred to as soft handoff. Soft handoff provides a dual diversity for the received signal from two links which improves the quality of reception and eliminates clicking as well as the ping-pong problem. Handoff is an important issue that has many more details and we will discuss these details in Chapter 6.

Power Control. As we discussed earlier in this chapter, power control is necessary for FDMA and TDMA systems to control adjacent channel interference and mitigate the unexpected interference caused by the near-far problem. In FDMA and TDMA systems, some sort of power control is needed to improve the quality of the voice delivered to the user. In CDMA, however, the capacity of the system depends *directly* on the power control, and an accurate power control mechanism is needed for proper operation of the network. With CDMA, power control is the key ingredient in maximizing the number of users that can operate simultaneously in the system. As a result, CDMA systems adjust the transmitted power more often and with smaller adjustment steps to support a more refined control of power. Better power control also saves on the transmission power of the MS, which increases the life of the battery. The more refined power control in CDMA systems also helps in power management of the MS, which is an extremely important practical issue for users of the mobile terminals. These issues are further discussed in Chapters 6 and 8.

Implementation Complexity. Spread spectrum is a two-layer modulation technique requiring greater circuit complexity than conventional modulation schemes.

This in turn will lead to higher electronic power consumption and larger weight and cost for mobile terminals. Gradual improvements in battery and integrated circuit technologies, however, have made this issue transparent to the user.

4.2.5 Performance of Fixed-Assignment Access Methods

Fixed assignment access methods are used with circuit switched cellular and PCS telephone networks. In these networks, in a manner similar to the wired multichannel environments, the performance of the network is measured by the blockage rate of an initiated call. A call does not go through for two reasons: (1) when the calling number is not available, and (2) when the telephone company is out of resources to provide a line for the communication session. In POTS, for both cases the user hears a busy tone signal and cannot distinguish between the two types of blockage. In most cellular systems, however, type (1) blockage results in a response that is a busy tone and type (2) with a message such as "All the circuits are busy at this time please try your call later." In the rest of this book, we refer to blockage rate only as a type (2) blockage rate. The statistical properties of the traffic offered to the network are also a function of time. The telephone service providers often design their networks so that the blockage rate at peak traffic is always below a certain percentage. Cellular operators often try to keep this average blockage rate below 2 percent.

The blockage rate is a function of the number of subscribers, number of initiated calls, and the length of the conversations. In telephone networks, the Erlang equations are used to relate the probability of blockage to the average rate of the arriving calls and the average length of a call. In wired networks, the number of lines or subscribers that can connect to a multichannel switch is a fixed number. The telephone company monitors the statistics of the calls over a long period of time and upgrades the switches with the growth of subscribers so that the blockage rate during peak traffic times remains below the objective value. In cellular telephony and PCS networks, the number of subscribers operating in a cell is also a function of time. In the downtown areas, everyone uses their cellular telephones during the day, and in the evenings they use them in their residential area which is covered by a different cell. Therefore, traffic fluctuations in cellular telephone networks are much more than the traffic fluctuations in POTS. In addition, telephone companies can easily increase the capacity of their networks by increasing their investment on the number of transmission lines and quality of switches supporting network connections. In wireless networks, the overall number of available channels for communications is ultimately limited by the availability of the frequency bands assigned for network operation. To respond to the fluctuations of the traffic and cope with the bandwidth limitations, cellular operators use complex frequency assignment strategies to share the available resources in an optimal manner. Some of these issues are discussed in Chapter 5.

4.2.5.1 Traffic Engineering Using the Erlang Equations

The Erlang equations are the core of the traffic engineering for telephony applications. The two basic equations used for traffic engineering are Erlang B and Erlang C equations. The Erlang B equation relates the probability of blockage

$B(N,\rho)$ to the number of channels N and the normalized call density in units of channels ρ. The Erlang B formula is:

$$B(N,\rho) = \frac{\rho^N / N!}{\sum_i^N (\rho^i / i!)} \tag{4.6}$$

where $\rho = \lambda / \mu$, λ is the call arrival rate and μ is the service rate of the calls.[1]

Problem 6: Call Blocking Using Erlang B Formula

We want to provide a wireless public phone service with five lines to a ferry crossing between Helsinki and Stockholm carrying 100 passengers where on the average each passenger makes a three-minute telephone call every two hours. What is the probability of a passenger approaching the telephones and none of the four lines are available?

Solution:

In practice, often the probability of call blockage is given, and we need to calculate the number of subscribers. Here we need an inverse function for the Erlang equation that is not available. As a result, a number of tables and graphs are available for this inverse mapping. Figure 4.8 shows a graph relating the probability of blockage $B(N,\rho)$ to the number of channels N and the normalized traffic per available channels ρ. From this graph, we can estimate the blocking probability. The traffic load is 100 users $\times$ 1 call/user $\times$ 3 minutes/call per 120 minutes = 2.5 Erlangs. Because there are five lines available and the traffic is 2.5 Erlangs, the blocking probability is roughly 0.07.

Problem 7: Capacity Using Erlang B Formula

An IS-136 cellular phone provider owns 50 cell sites and 19 traffic carriers per cell each with a bandwidth of 30 kHz. Assuming each user makes three calls per hour and the average holding time per call of five minutes, determine the total number of subscribers that the service provider can support with a blocking rate of less than 2 percent.

Solution:

The total number of channels is $N = 19 \times 3 = 57$ per cell. For $B(N,\rho) = 0.02$ and $N = 57$ Figure 4.8 shows that the $\rho = 45$ Erlangs. With an average of five calls per minute, the service rate is $\mu = 1/5$ minutes, and the acceptable arrival rate of the calls is $\lambda = \rho \times \mu = 1/5$ (min^{-1}) $\times$ 45 (Erlang) = 9 (Erlang/min). With an average of 3 calls per hour, the system can accept 9 (Erlang/min) / 3 (Erlang) / 60 (min) = 180 subscribers per cell. Therefore the total number of subscribers are 180 (subscribers/cell) $\times$ 50 (cells) = 8,000 subscribers.

The Erlang C formula relates the waiting time in a queue if a call does not go through, but it is buffered until a channel is available. These equations start with

[1] The equation assumes that the arrivals are Poisson, and the service rate is exponential. For details, see [BER87].