CHAPTER 6

Multiple Access Techniques

6.1 Introduction

In Chapter 5 we discussed that cellular systems divide a geographic region into cells where mobile units in each cell communicate with the cell's base station. The goal in the design of a cellular system is to be able to handle as many calls as possible in a given bandwidth with the specified blocking probability (reliability).

Multiplexing deals with the division of the resources to create multiple channels. Multiplexing can create channels in frequency, time, etc., and the corresponding terms are then frequency division multiplexing (FDM), time division multiplexing (TDM), etc. [1,3]. Since the amount of spectrum available is limited, we need to find ways to allow multiple users to share the available spectrum simultaneously. Shared access is used to implement a multiple access scheme when access by many users to a channel is required [13,14,15]. For example, one can create multiple channels using TDM, but each of these channels can be accessed by a group of users using the ALOHA multiple access scheme [8,9]. The multiple access schemes can be either reservation-based or random.

Multiple access schemes allow many users to share the radio spectrum. Sharing the bandwidth efficiently among users is one of the main objectives of multiple access schemes [16,17]. The variability of wireless channels presents both challenges and opportunities in designing multiple access communications systems. Multiple access strategy has an impact on robustness and interference levels generated in other cells. Therefore, multiple access schemes are designed to maintain orthogonality and reduce interference effects [10].

Multiple access schemes can be classified as *reservation-based* multiple access (e.g., FDMA, TDMA, CDMA) [4,5] and *random* multiple access (e.g., ALOHA, CSMA) (see Figure 6.1) [9,23]. If data traffic is continuous and a small transmission delay is required (for example in voice communication) reservation-based multiple access is used. The family of reservation-based multiple access includes frequency division multiple access (FDMA), time division multiple access (TDMA), and code division multiple access (CDMA) [6,7,12,21,22]. In many wireless systems for voice communication, the control channel is based on random multiple access and the communication channel is based on FDMA, TDMA, or CDMA. The reservation-based multiple access technique has a disadvantage in that once the channel is assigned, it remains idle if the user has nothing to transmit,

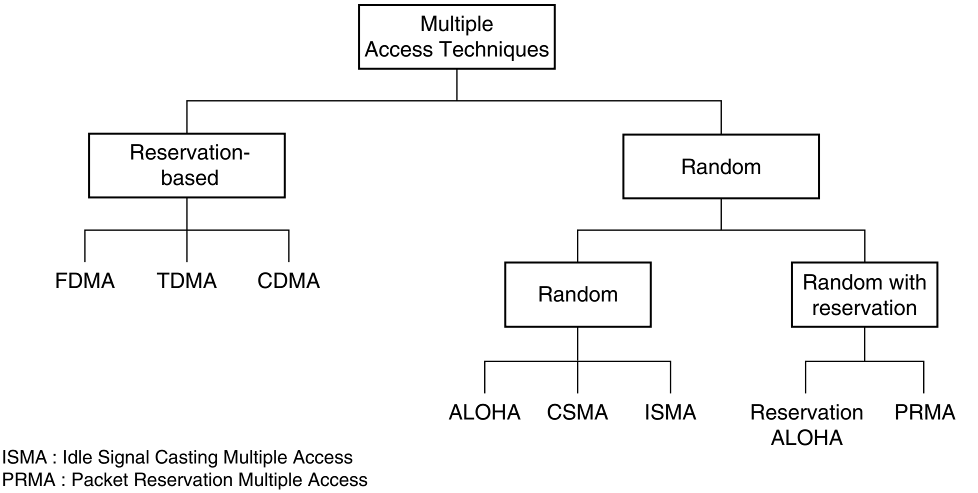

Figure 6.1 Multiple access schemes.

while other users may have data waiting to be transmitted. This problem is critical when data generation is random and has a high peak-rate to average-rate ratio. In this situation, random multiple access is more efficient, because a communication channel is shared by many users and users transmit their data in a random or partially coordinated fashion. ALOHA and carrier sense multiple access (CSMA) are examples of random multiple access [8]. If the data arrives in a random manner, and the data length is large, then random multiple access combined with a reservation protocol will perform better than both random- and reservation-based schemes.

We first focus on the reservation-based multiple access schemes including narrowband channelized and wideband nonchannelized systems for wireless communications. We discuss access technologies—FDMA, TDMA, and CDMA. We examine FDMA and TDMA from a capacity, performance, and spectral efficiency viewpoint. As networks have evolved, the demand for higher capacities has encouraged researchers and system designers to examine access schemes that are even more spectrally efficient than TDMA. Therefore, we also examine the CDMA system. Work in standards bodies around the world indicates that the 3G/4G wireless systems are evolving to wideband CDMA (WCDMA) to achieve high efficiencies and high access data rates. The later part of the chapter is devoted to the discussion of random multiple access schemes.

6.2 Narrowband Channelized Systems

Traditional architectures for analog and digital wireless systems are channelized [6,11]. In a channelized system, the total spectrum is divided into a large number of relatively narrow radio channels that are defined by carrier frequency. Each radio channel consists of a pair of frequencies. The frequency used for transmission from the base station to the mobile station is called the *forward channel* (downlink channel) and the frequency used for transmission from the mobile station to the base station is called the *reverse channel* (uplink channel). A user is assigned both frequencies for the duration of the call. The forward and reverse channels are assigned widely separated frequencies to keep the interference between transmission and reception to a minimum.

A narrowband channelized system demands precise control of output frequencies for an individual transmitter. In this case, the transmission by a given mobile station occurs within the specified narrow bandwidth to avoid interference with adjacent channels. The tightness of bandwidth limitations plays a dominant role in the evaluation and selection of modulation technique. It also influences the design of transmitter and receiver elements, particularly the filters which can greatly affect the cost of a mobile station.

A critical issue with regulators and operators around the world is how efficiently the radio spectrum is being used. Regulatory bodies want to encourage competition for cellular services. Thus for a given availability of bandwidth, more operators can be licensed. For a particular operator, a more efficient technology can support more users within the assigned spectrum and thus increase profits.

When we examine efficiencies of various technologies, we find that each system has made different trade-offs in determining the optimum method for access. Some of the parameters that are used in the trade-off are bandwidth per user, guard bands between channels, frequency reuse among different cells in the system, the signal-to-noise and signal-to-interference ratio, the methods of channel and speech coding, and the complexity of the system.

The first-generation analog cellular systems showed signs of capacity saturation in major urban areas, even with a modest total user population. A major capacity increase was needed to meet future demand. Several digital techniques were deployed to solve the capacity problem of analog cellular systems. There are two basic types of systems whereby a fixed spectrum resource is partitioned and shared among different users [13,16]. The channels are created by dividing the total system bandwidth into frequency channels through the use of FDM and then further dividing each frequency channel into time channels through the use of TDM. Most systems use a combination of FDMA and TDMA.

6.2.1 Frequency Division Duplex (FDD) and Time Division Duplex (TDD) System

Many cellular systems (such as AMP, GSM, DAMP, etc.) use *frequency division duplex* (*FDD*) in which the transmitter and receiver operate simultaneously on different frequencies. Separation is provided between the downlink and uplink channels to avoid the transmitter causing self interference to its receiver. Other precautions are also needed to prevent self interference, such as the use of two antennas, or alternatively one antenna with a duplexer (a special design of RF filters protecting the receiver from the transmit frequency). A duplexer adds weight, size, and cost to a radio transceiver and can limit the minimum size of a subscriber unit.

A cellular system can be designed to use one frequency band by using *time division duplex* (*TDD*). In TDD a bidirectional flow of information is achieved using the simplex-type scheme by automatically alternating in time the direction of transmission on a single frequency. At best TDD can only provide a quasi-simultaneous bidirectional flow, since one direction must be off while the other is using the frequency. However, with a high enough transmission rate on the channel, the off time is not noticeable during conversations, and with a digital speech system, the only effect is a very short delay.

The amount of spectrum required for both FDD and TDD is the same. The difference lies in the use of two bands of spectrum separated by the required bandwidth for FDD, whereas TDD requires only one band of frequencies but twice the bandwidth. It may be easier to find a single band of unassigned frequencies than finding two bands separated by the required bandwidth.

With TDD systems, the transmit time slot and the receiver time slot of the subscriber unit occur at different times. With the use of a simple RF switch in the subscriber unit, the antenna can be connected to the transmitter when a transmit burst is required (thus disconnecting the receiver from the antenna) and to the receiver for the incoming signal. The RF switch thus performs the function of the duplexer, but is less complex, smaller in size, and less costly. TDD uses a burst mode scheme like TDMA and therefore also does not require a duplexer. Since the bandwidth of a TDD channel is twice that of a transmitter and receiver in an FDD system, RF filters in all the transmitters and receivers for TDD systems must be designed to cover twice the bandwidth of FDD system filters.

6.2.2 Frequency Division Multiple Access

The FDMA is the simplest scheme used to provide multiple access. It separates different users by assigning a different carrier frequency (see Figure 6.2). Multiple users are isolated using bandpass filters. In FDMA, signals from various users are assigned different frequencies, just as in an analog system. Frequency guard bands are provided between adjacent signal spectra to minimize crosstalk between adjacent channels.

The advantages and disadvantages of FDMA with respect to TDMA or CDMA are:

Advantages

1. Capacity can be increased by reducing the information bit rate and using an efficient digital speech coding scheme (See Chapter 8) [20].

2. Technological advances required for implementation are simple. A system can be configured so that improvements in terms of a lower bit rate speech coding could be easily incorporated.
3. Hardware simplicity, because multiple users are isolated by employing simple bandpass filters.

Disadvantages

1. The system architecture based on FDMA was implemented in first-generation analog systems such as advanced mobile phone system (AMPS) or total access communication system (TACS). The improvement in capacity depends on operation at a reduced signal-to-interference (*S/I*) ratio. But the narrowband digital approach gives only limited advantages in this regard so that modest capacity improvements could be expected from the allocated spectrum.
2. The maximum bit-rate per channel is fixed and small, inhibiting the flexibility in bit-rate capability that may be a requirement for computer file transfer in some applications in the future.

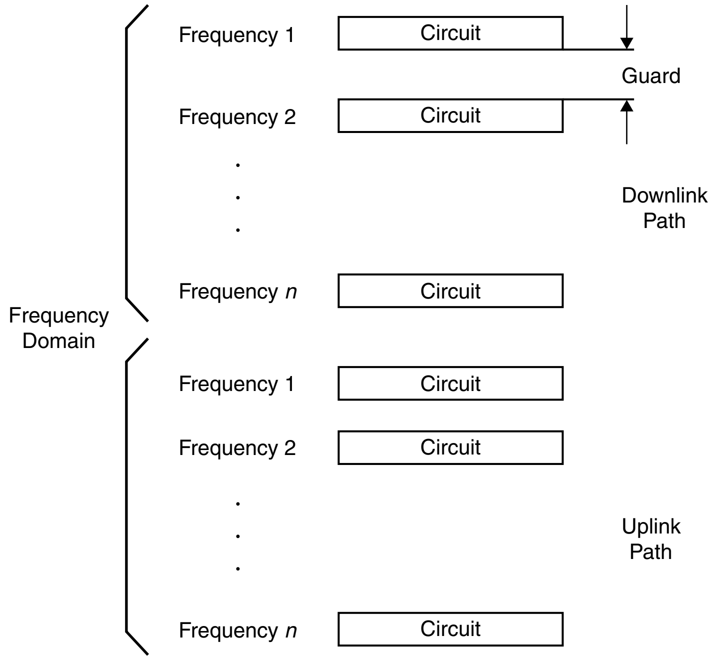

Figure 6.2 FDMA/FDD channel architecture.

3. Inefficient use of spectrum, in FDMA if a channel is not in use, it remains idle and cannot be used to enhance the system capacity.
4. Crosstalk arising from adjacent channel interference is produced by nonlinear effects.

6.2.3 Time Division Multiple Access

In a TDMA system, each user uses the whole channel bandwidth for a fraction of time (see Figure 6.3) compared to an FDMA system where a single user occupies the channel bandwidth for the entire duration (see Figure 6.2) [2]. In a TDMA system, time is divided into equal time intervals, called *slots*. User data is transmitted in the slots. Several slots make up a frame. Guard times are used between each user's transmission to minimize crosstalk between channels (see Figure 6.4). Each user is assigned a frequency and a time slot to transmit data. The data is transmitted via a radio-carrier from a base station to several active mobiles in the downlink. In the reverse direction (uplink), transmission from mobiles to base stations is time-sequenced and synchronized on a common frequency for TDMA. The preamble carries the address and synchronization information that both the base station and mobile stations use for identification.

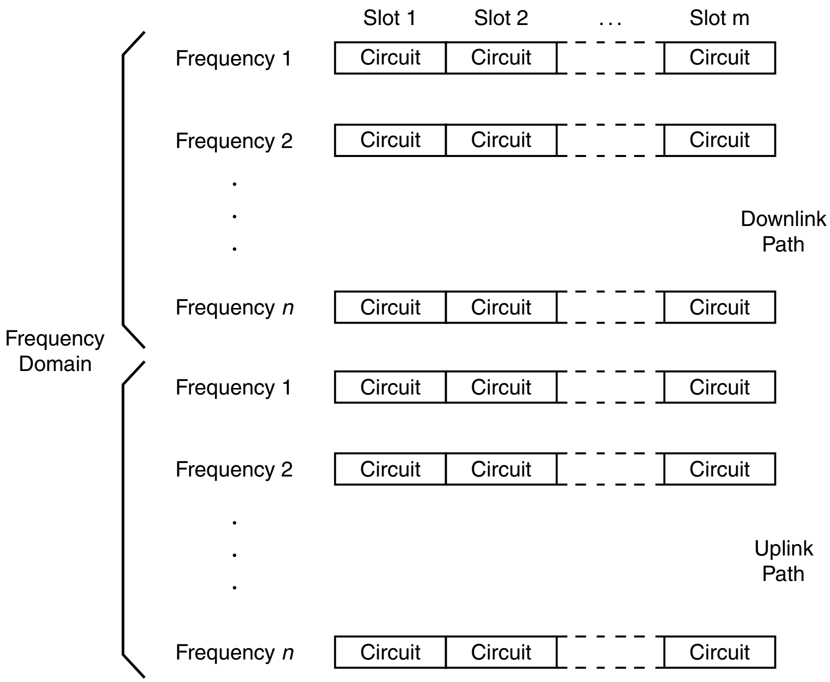

Figure 6.3 TDMA/FDD channel architecture.

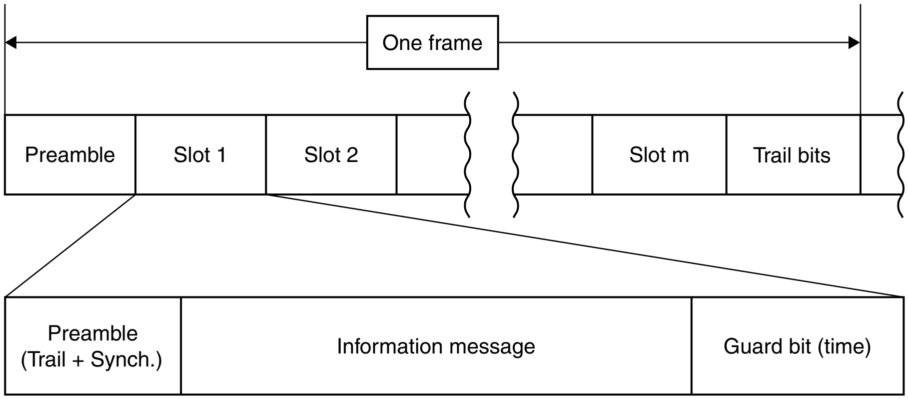

Figure 6.4 TDMA frame.

In a TDMA system, the user can use multiple slots to support a wide range of bit rates by selecting the lowest multiplexing rate or multiple of it. This enables supporting a variety of voice coding techniques at different bit rates with different voice qualities. Similarly, data communications customers could use the same kinds of schemes, choosing and paying for the digital data rate as required. This would allow customers to request and pay for a bandwidth on demand.

Depending on the data rate used and the number of slots per frame, a TDMA system can use the entire bandwidth of the system or can employ an FDD scheme. The resultant multiplexing is a mixture of frequency division and time division. The entire frequency band is divided into a number of duplex channels (about 350 to 400 kHz). These channels are deployed in a frequency-reuse pattern, in which radio-port frequencies are assigned using an autonomous adaptive frequency assignment algorithm. Each channel is configured in a TDM mode for the downlink direction and a TDMA mode for the uplink direction.

The advantages and disadvantages of TDMA are:

Advantages

1. TDMA permits a flexible bit rate, not only for multiples of the basic single channel rate but also submultiples for low bit rate broadcast-type traffic.
2. TDMA offers the opportunity for frame-by-frame monitoring of signal strength/bit error rates to enable either mobiles or base stations to initiate and execute handoffs.
3. TDMA, when used exclusively and not with FDMA, utilizes bandwidth more efficiently because no frequency guard band is required between channels.

4. TDMA transmits each signal with sufficient guard time between time slots to accommodate time inaccuracies because of clock instability, delay spread, transmission delay because of propagation distance, and the tails of signal pulse because of transient responses.

Disadvantages

1. For mobiles and particularly for hand-sets, TDMA on the uplink demands high peak power in transmit mode, that shortens battery life.
2. TDMA requires a substantial amount of signal processing for matched filtering and correlation detection for synchronizing with a time slot.
3. TDMA requires synchronization. If the time slot synchronization is lost, the channels may collide with each other.
4. One complicating feature in a TDMA system is that the propagation time for a signal from a mobile station to a base station varies with its distance to the base station.

6.3 Spectral Efficiency

An efficient use of the spectrum is the most desirable feature of a mobile communications system. To realize this, a number of techniques have been proposed or already implemented. Some of these techniques used to improve spectral efficiency are reducing the channel bandwidth, information compression (low-rate speech coding), variable bit rate codec (see Chapter 8), improved channel assignment algorithms (dynamic channel assignment), and so on [11,17,19]. Spectral efficiency of a mobile communications system shows how efficiently the spectrum is used by the system. Spectral efficiency of a mobile communications system depends on the choice of a multiple access scheme. The measure of spectral efficiency enables one to estimate the capacity of a mobile communications system.

The overall spectral efficiency of a mobile communications system can be estimated by knowing the *modulation* and the *multiple access* spectral efficiencies separately [16].

6.3.1 Spectral Efficiency of Modulation

The spectral efficiency with respect to modulation is defined as [16]:

$$\eta_m = \frac{\text{(Total Number of Channels Available in the System)}}{\text{(Bandwidth)(Total Coverage Area)}} \tag{6.1a}$$

$$\eta_m = \frac{\dfrac{B_w}{B_c} \times \dfrac{N_c}{N}}{B_w \times N_c \times A_c} \tag{6.1b}$$

$$\eta_m = \frac{1}{B_c \times N \times A_c} \text{ Channels/MHz/km}^2 \tag{6.1c}$$

where:

η_m = modulation efficiency (channels/MHz/km^2)
B_w = bandwidth of the system (MHz)
B_c = channel spacing (MHz)
N_c = total number of cells in the covered area
N = frequency reuse factor of the system (or cluster size)
A_c = area covered by a cell (km^2)

Equation 6.1c indicates that the spectral efficiency of modulation does not depend on the bandwidth of the system. It only depends on the channel spacing, the cell area, and the frequency reuse factor, N. By reducing the channel spacing, the spectral efficiency of modulation for the system is increased, provided the cell area (A_c) and reuse factor (N) remain unchanged. If a modulation scheme can be designed to reduce N then more channels are available in a cell and efficiency is improved.

Another definition of spectral efficiency of modulation is *Erlangs/MHz/km*^{2}

$$\eta_m = \frac{\text{Maximum Total Traffic Carried by System}}{\text{(System Bandwidth)(Total Coverage Area)}} \tag{6.2a}$$

$$\eta_m = \frac{\text{Total Traffic Carried by} \left(\frac{B_w/B_c}{N}\right) \text{Channels}}{B_w A_c} \tag{6.2b}$$

By introducing the trunking efficiency factor, η_t in Equation 6.2b (<1, it is a function of the blocking probability and number of available channels per cell), the total traffic carried by the system is given as:

$$\eta_m = \frac{\eta_t \left(\frac{B_w/B_c}{N}\right)}{B_w A_c} \tag{6.2c}$$

$$\eta_m = \frac{\eta_t}{B_c N A_c} \tag{6.2d}$$

where:

η_t is a function of the blocking probability and the total number of available channels per cell $\left[\frac{B_w/B_c}{N}\right]$

Based on Equation 6.2d we can conclude:

1. The voice quality will depend on the frequency reuse factor, N, which is a function of the signal-to-interference (S/I) ratio of the modulation scheme used in the mobile communications system (see Chapter 5).
2. The relationship between system bandwidth, B_w, and the amount of traffic carried by the system is nonlinear, i.e., for a given percentage increase in B_w, the increase in the traffic carried by the system is more than the increase in B_w.
3. From the average traffic per user (*Erlang/user*) during the busy hour and *Erlang/MHz/km*^{2}, the capacity of the system in terms of *users/km*^{2}*/MHz* can be obtained.
4. The spectral efficiency of modulation depends on the blocking probability.

Example 6.1

In the GSM800 digital channelized cellular system, the one-way bandwidth of the system is 12.5 MHz. The RF channel spacing is 200 kHz. Eight users share each RF channel and three channels per cell are used for control channels. Calculate the spectral efficiency of modulation (for a dense metropolitan area with small cells) using the following parameters:

- Area of a cell = 8 km^2
- Total coverage area = 4000 km^2
- Average number of calls per user during the busy hour = 1.2
- Average holding time of a call = 100 seconds
- Call blocking probability = 2%
- Frequency reuse factor = 4

Solution

$$\text{Number of 200 kHz RF channels} = \frac{12.5 \times 1000}{200} = 62$$

$$\text{Number of traffic channels} = 62 \times 8 = 496$$

$$\text{Number of signaling channels per cell} = 3$$

$$\text{Number of traffic channels per cell} = \frac{496}{4} - 3 = 121$$

$$\text{Number of cells} = \frac{4000}{8} = 500$$

With 2% blocking for an omnidirectional case, the total traffic carried by 121 channels (using Erlang-B tables) = 108.4 (1.0 − 0.02) = 106.2 Erlangs/cell or 13.28 Erlangs/km^2

Number of calls per hour per cell $= \frac{106.2 \times 3600}{100} = 3823$, calls/hour/km^2 $= \frac{3823}{8} = 477.9$ calls/hour/km^2

Max. number of users/cell/hour $= \frac{3823}{1.2} = 3186$, users/hour/channel $= \frac{3186}{121} = 26.33$

$$\eta_m = \frac{(\text{Erlangs per cell}) \times \text{no. of cells}}{B_w \times \text{Coverage Area}} = \frac{106.2 \times 500}{12.5 \times 4000} = 1.06 \, \text{Erlangs/MHz/km}^2$$

6.3.2 Multiple Access Spectral Efficiency

Multiple access spectral efficiency is defined as the ratio of the total time or frequency dedicated for traffic transmission to the total time or frequency available to the system. Thus, the multiple access spectral efficiency is a dimensionless number with an upper limit of unity.

In FDMA, users share the radio spectrum in the frequency domain. In FDMA, the multiple access efficiency is reduced because of guard bands between channels and also because of signaling channels. In TDMA, the efficiency is reduced because of guard time and synchronization sequence.

FDMA Spectral Efficiency

For FDMA, multiple access spectral efficiency is given as:

$$\eta_a = \frac{B_c N_T}{B_w} \leq 1 \tag{6.3}$$

where:

η_a = multiple access spectral efficiency
N_T = total number of traffic channels in the covered area
B_c = channel spacing
B_w = system bandwidth

Example 6.2

In a first-generation AMP system where there are 395 channels of 30 kHz each in a bandwidth of 12.5 MHz, what is the multiple access spectral efficiency for FDMA?

Solution

$$\eta_a = \frac{30 \times 395}{12.5 \times 1000} = 0.948$$

TDMA Spectral Efficiency

TDMA can operate as wideband or narrowband. In the wideband TDMA, the entire spectrum is used by each individual user. For the wideband TDMA, multiple access spectral efficiency is given as:

$$\eta_a = \frac{\tau M_t}{T_f} \tag{6.4}$$

where:

τ = duration of a time slot that carries data
T_f = frame duration
M_t = number of time slots per frame

In Equation 6.4 it is assumed that the total available bandwidth is shared by all users. For the narrowband TDMA schemes, the total band is divided into a number of sub-bands, each using the TDMA technique. For the narrowband TDMA system, frequency domain efficiency is not unity as the individual user channel does not use the whole frequency band available to the system. The multiple access spectral efficiency of the narrowband TDMA system is given as:

$$\eta_a = \left(\frac{(\tau M_t)}{T_f}\right)\left(\frac{(B_u N_u)}{B_w}\right) \tag{6.5}$$

where:

B_u = bandwidth of an individual user during his or her time slot
N_u = number of users sharing the same time slot in the system, but having access to different frequency sub-bands

6.3.3 Overall Spectral Efficiency of FDMA and TDMA Systems

The overall spectral efficiency, η, of a mobile communications system is obtained by considering both the modulation and multiple access spectral efficiencies

$$\eta = \eta_m \eta_a \tag{6.6}$$

Example 6.3

In the North American Narrowband TDMA cellular system, the one-way bandwidth of the system is 12.5 MHz. The channel spacing is 30 kHz and the total number of voice channels in the system is 395. The frame duration is 40 ms, with six time slots per frame. The system has an individual user data rate of 16.2 kbps in which the speech with error protection has a rate of 13 kbps. Calculate the multiple access spectral efficiency of the TDMA system.

Solution

The time slot duration that carries data: $\tau = \left(\frac{13}{16.2}\right)\left(\frac{40}{6}\right) = 5.35\,\text{ms}$

$T_f = 40\,\text{ms}$, $M_t = 6$, $N_u = 395$, $B_u = 30\,\text{kHz}$, and $B_w = 12.5\,\text{MHz}$

$$\eta_a = \frac{5.35 \times 6}{40} \times \frac{30 \times 395}{12500} = 0.76$$

The overhead portion of the frame $= 1.0 - 0.76 = 24\%$

Capacity and Frame Efficiency of a TDMA System

Cell Capacity

The cell capacity is defined as the maximum number of users that can be supported simultaneously in each cell.

The capacity of a TDMA system is given by [16]:

$$N_u = \frac{\eta_b \mu}{\nu_f} \times \frac{B_w}{RN} \tag{6.7}$$

where:

N_u = number of channels (mobile users) per cell
η_b = bandwidth efficiency factor (<1.0)
μ = bit efficiency (= 2 bit/symbol for QPSK, = 1 bit/symbol for GMSK as used in GSM)
ν_f = voice activity factor (equal to one for TDMA)
B_w = one-way bandwidth of the system
R = information (bit rate plus overhead) per user
N = frequency reuse factor

$$\text{Spectral efficiency } \eta = \frac{N_u \times R}{B_w} \text{ bit/sec/Hz} \tag{6.8}$$

Example 6.4

Calculate the capacity and spectral efficiency of a TDMA system using the following parameters: bandwidth efficiency factor $\eta_b = 0.9$, bit efficiency (with QPSK) $\mu = 2$, voice activity factor $\nu_f = 1.0$, one-way system bandwidth $B_w = 12.5\,\text{MHz}$, information bit rate $R = 16.2\,\text{kbps}$, and frequency reuse factor $N = 19$.

Solution

$$N_u = \frac{0.9 \times 2}{1.0} \times \frac{12.5 \times 10^6}{16.2 \times 10^3 \times 19}$$

$N = 73.1$ (say 73 mobile users per cell)

$$\text{Spectral efficiency } \eta = \frac{73 \times 16.2}{12.5 \times 1000} = 0.094 \text{ bit/sec/Hz}$$

Efficiency of a TDMA Frame

We refer to Figure 6.4 that shows a TDMA frame. The number of overhead bits per frame is:

$$b_0 = N_r b_r + N_t b_p + (N_t + N_r) b_g \tag{6.9}$$

where:

N_r = number of reference bursts per frame
N_t = number of traffic bursts (slots) per frame
b_r = number of overhead bits per reference burst
b_p = number of overhead bits per preamble per slot
b_g = number of equivalent bits in each guard time interval

The total number of bits per frame is:

$$b_T = T_f \times R_{rf} \tag{6.10a}$$

where:

T_f = frame duration
R_{rf} = bit rate of the RF channel

$$\text{Frame efficiency } \eta = (1 - b_0/b_T) \times 100\% \tag{6.10b}$$

It is desirable to maintain the efficiency of the frame as high as possible.

The number of bits per data channel (user) per frame is $b_c = RT_f$, where R = bit rate of each channel (user).

$$\text{No. of channels/frame} \qquad N_{CF} = \frac{\text{(Total Data Bits)/(frame)}}{\text{(Bits per Channel)/(frame)}}$$

$$N_{CF} = \frac{\eta R_{rf} T_f}{R T_f} \tag{6.11a}$$

$$N_{CF} = \frac{\eta R_{rf}}{R} \tag{6.11b}$$

Equation 6.11b indicates the number of time slots per frame.

Example 6.5

Consider the GSM TDMA system with the following parameters:

$N_r = 2$
$N_t = 24$ frames of 120 ms each with eight time slots per frame
$b_r = 148$ bits in each of 8 time slots
$b_p = 34$ bits in each of 8 time slots
$b_g = 8.25$ bits in each of 8 time slots
$T_f = 120$ ms
$R_{rf} = 270.8333333$ kbps
$R = 22.8$ kbps

Calculate the frame efficiency and the number of channels per frame.

Solution

$$b_0 = 2 \times (8 \times 148) + 24 \times (8 \times 34) + 8 \times 8.25 = 10{,}612 \text{ bits per frame}$$

$$b_T = 120 \times 10^{-3} \times 270.8333333 \times 10^3 = 32{,}500 \text{ bits per frame}$$

$$\eta = \left(1 - \frac{10612}{32500}\right) \times 100 = 67.35\%$$

$$\text{Number of channels/frame} = \frac{0.6735 \times 270.8333333}{22.8} = 8$$

The last calculation, with an answer of 8 channels, confirms that our calculation of efficiency is correct.

6.4 Wideband Systems

In wideband systems, the entire system bandwidth is made available to each user, and is many times larger than the bandwidth required to transmit information. Such systems are known as *spread spectrum* (SS) systems. There are two fundamental types of spread spectrum systems: (1) direct sequence spread spectrum (DSSS) and (2) frequency hopping spread spectrum (FHSS) [3,26].

In a DSSS system, the bandwidth of the baseband information carrying signals from a different user is spread by different codes with a bandwidth much larger than that of the baseband signals (see Chapter 11 for details). The spreading codes used for different users are orthogonal or nearly orthogonal to each other. In the DSSS, the spectrum of the transmitted signal is much wider than the spectrum associated with the information rate. At the receiver, the same code is used for despreading to recover the baseband signal from the target user while suppressing the transmissions from all other users (see Figure 6.5).

One of the advantages of the DSSS system is that the transmission bandwidth exceeds the coherence bandwidth (see Chapter 3). The received signal, after despreading (see Chapter 11 for details), resolves into multiple signals with different time delays. A Rake receiver (see Chapter 11) can be used to recover the multiple time

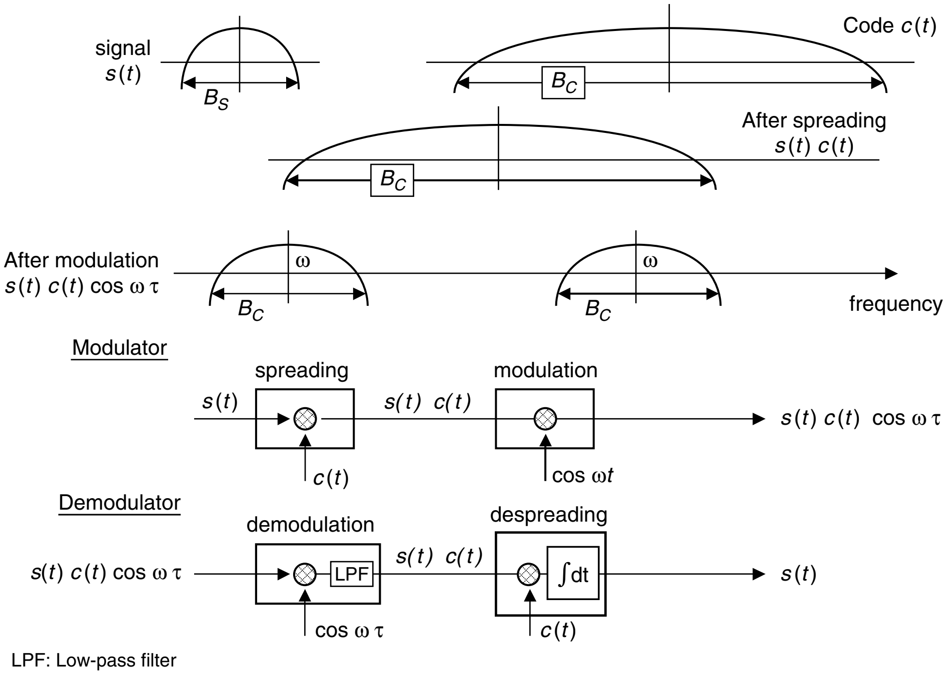

Figure 6.5 Direct sequence spread spectrum.

delayed signals and combine them into one signal to provide a time diversity with a lower frequency of deep fades. Thus, the DSSS system provides an inherent robustness against mobile-channel degradations. Another potential benefit of a DSSS system is the greater resistance to interference effects in a frequency reuse situation. Also, there may be no hard limit on the number of mobile users who can simultaneously gain access. The capacity of a DSSS system depends upon the desired value of E_b/I_0 instead of resources (frequencies or time slots) as in FDMA or TDMA systems.

Frequency hopping (FH) is the periodic changing of the frequency or the frequency set associated with transmission (see Figure 6.6). If the modulation is M-*ary* frequency-shift-keying (MFSK) (see Chapter 9 for details), two or more frequencies are in the set that change at each hop. For other modulations, a single center or carrier frequency is changed at each hop.

An FH signal may be considered a sequence of modulated pulses with pseudo-random carrier frequencies. The set of possible carrier frequencies is called the hop set. Hopping occurs over a frequency band that includes a number of frequency channels. The bandwidth of a frequency channel is called the *instantaneous bandwidth* (B_I). The bandwidth of the frequency band over which the hopping occurs is called the total *hopping bandwidth* (B_H). The time duration between hops is called the *hop duration* or hopping period (T_H).

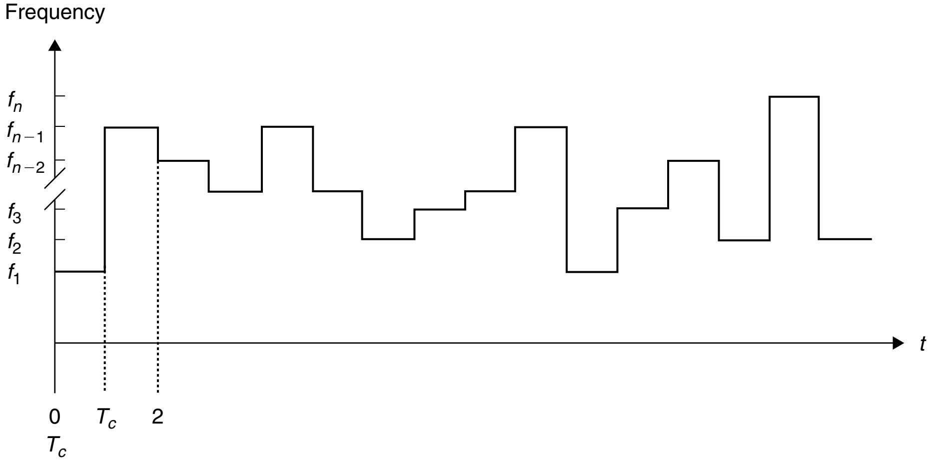

Figure 6.6 Frequency hopping spread spectrum system.

Frequency hopping can be classified as fast or slow. *Fast frequency hopping* occurs if there is frequency hop for each transmitted symbol. Thus, fast frequency hopping implies that the hopping rate equals or exceeds the information symbol rate. *Slow frequency hopping* occurs if two or more symbols are transmitted in the time interval between frequency hops.

Frequency hopping allows communicators to hop out of frequency channels with interference or to hop out of fades. To exploit this capability, error-correcting codes, appropriate interleaving, and disjoint frequency channels are nearly always used. A frequency synthesizer is required for frequency hopping systems to convert a stable reference frequency into the various frequency of hop set.

Frequency hopping communicators do not often operate in isolation. Instead, they are usually elements of a network of frequency hopping systems that create mutual multiple-access interference. This network is called a frequency-hopping multiple-access (FHMA) network.

If the hoppers of an FHMA network all use the same M frequency channels, but coordinate their frequency transitions and their hopping sequence, then the multiple-access interference for a lightly loaded system can be greatly reduced compared to a non-hopped system. For the number of hopped signals (M_h) less than the number of channels (N_c), a coordinated hopping pattern can eliminate interference. As the number of hopped signals increases beyond N_c, then the interference will increase in proportion to the ratio of the number of signals to the number of channels. In the absence of fading or multipath interference, since there is no interference suppression system in frequency hopping, for a high channel loading the performance of a frequency hopping system is no better than a non-hopped system. Frequency hopping systems are best for light channel loadings in the presence of conventional non-hopped systems.

When fading or multipath interference is present, the frequency hopping system has better error performance than a non-hopped system. If the transmitter hops to a channel in a fade, the errors are limited in duration since the system will shortly hop to a new frequency where the fade may not be as deep.

Network coordination for frequency hopping systems are simpler to implement than that for DSSS systems because the timing alignments must be within a fraction of a hop duration, rather than a fraction of a sequence chip (narrow pulse). In general, frequency hopping systems reject interference by trying to avoid it, whereas DSSS systems reject interference by spreading it. The interleaving and error-correcting codes that are effective with frequency hopping systems are also effective with DSSS systems.

The major problems with frequency hopping systems with increasing hopping rates are the cost of the frequency synthesizer increases and its reliability decreases, and synchronization becomes more difficult.

In theory, a wideband system can be overlaid on existing, fully loaded, narrowband channelized systems (as an example, the IS-95 CDMA system overlays on existing AMPS [FDMA]). Thus, it may be possible to create a wideband network right on top of the narrowband cellular system using the same spectrum.

6.5 Comparisons of FDMA, TDMA, and DS-CDMA

The DSSS approach is the basis to implementation of the direct sequence code division multiple access (DS-CDMA) technique introduced by Qualcom. The DS-CDMA has been used in commercial applications of mobile communications. The primary advantage of DS-CDMA is its ability to tolerate a fair amount of interfering signals compared to FDMA and TDMA that typically cannot tolerate any such interference(Figure 6.7). As a result of the interference tolerance of CDMA, the problems of frequency band assignment and adjacent cell interference are greatly simplified. Also, flexibility in system design and deployment are significantly improved since interference to others is not a problem. On the other hand, FDMA and TDMA radios must be carefully assigned a frequency or time slot to assure that there is no interference with other similar radios. Therefore, sophisticated filtering and guard band protection is needed with FDMA and TDMA technologies. With DS-CDMA, adjacent microcells share the same frequencies whereas with FDMA/TDMA it is not feasible for adjacent microcells to share the same frequencies because of interference. In both FDMA and TDMA systems, a time-consuming frequency planning task is required whenever a network changes, whereas no such frequency planning is needed for a CDMA network since each cell uses the same frequencies.

Capacity improvements with DS-CDMA also result from voice activity patterns during two-way conversations, (i.e., times when a party is not talking) that cannot be cost-effectively exploited in FDMA or TDMA systems. DS-CDMA radios can, therefore, accommodate more mobile users than FDMA/TDMA radios

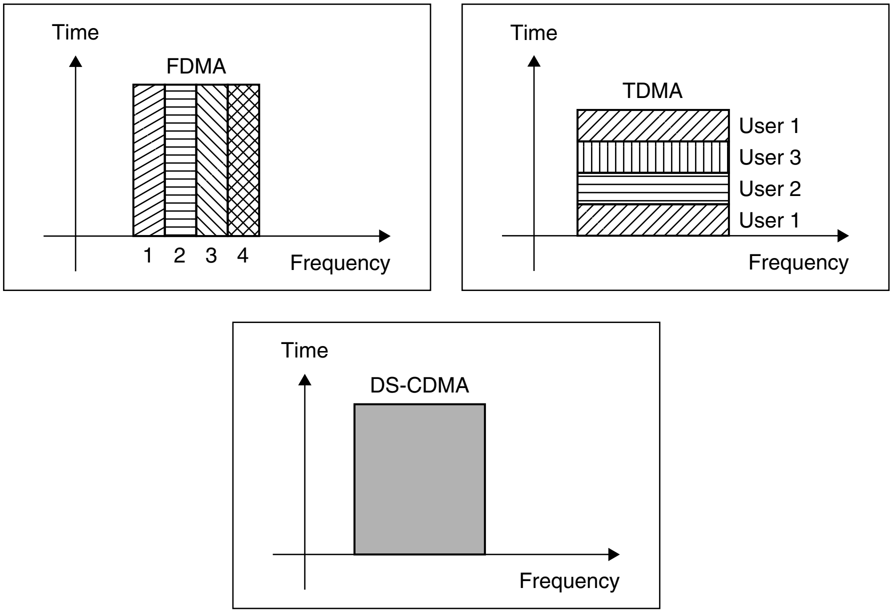

Figure 6.7 Comparison of multiple access methods.

on the same bandwidth. Further capacity gains for FDMA, TDMA, and CDMA can also result from antenna technology advancement by using directional antennas that allow the microcell area to be divided into sectors. Table 6.1 provides a summary of access technologies used for various wireless systems.

Table 6.1 Access technologies for wireless system.

System	Access technology	Mode of operation	Frame rate (kbps)
North American IS-54 (Dual Mode)	TDMA/FDD FDMA/FDD	Digital/ Analog FM	48.6 —
North American IS-95 (Dual Mode)	DS-CDMA/FDD FDMA/FDD	Digital/ Analog FM	1228.8 —
North American IS-136	TDMA/FDD	Digital	48.6
GSM (used all over world)	TDMA/FDD	Digital	270.833
European CT-2 Cordless	FDMA/TDD	Digital	72.0
DECT Cordless	TDMA/TDD	Digital	1152.0

6.6 Capacity of a DS-CDMA System

The capacity of a DS-CDMA system depends on the processing gain, G_p (a ratio of spreading bandwidth, B_w, and information rate, R), the bit energy-to-interference ratio, E_b/I_0, the voice duty cycle, v_f, the DS-CDMA omnidirectional frequency reuse efficiency, η_f, and the number of sectors, G, in the cell-site antenna.

The received signal power at the cell from a mobile is $S = R \times E_b$. The signal-to-interference ratio is

$$\frac{S}{I} = \frac{R}{B_w} \times \frac{Eb}{I_0} \tag{6.12}$$

where:

E_b = energy per bit
I_0 = interference density

In a cell with N_u mobile transmitters, the number of effective interferers is $N_u - 1$ because each mobile is an interferer to all other mobiles. This is valid regardless of how the mobiles are distributed within the cell since automatic power control (APC) is used in the mobiles. The APC operates such that the received power at the cell from each mobile is the same as for every other mobile in the cell, regardless of the distance from the center of the cell. APC conserves battery power in the mobiles, minimizes interference to other users, and helps overcome fading.

In a hexagonal cell structure, because of interference from each tier, the S/I ratio is given as (see Chapter 5):

$$\frac{S}{I} = \frac{1}{(N_u - 1) \times [1 + 6 \times k_1 + 12 \times k_2 + 18 \times k_3 + \cdots]} \tag{6.13}$$

where:

N_u = number of mobile users in the band, B_w

k_i, $i = 1, 2, 3, \ldots$ = the interference contribution from all terminals in individual cells in tiers 1, 2, 3, etc., relative to the interference from the center cell. This loss contribution is a function of both the path loss to the center cell and the power reduction because of power control to an interfering mobile's own cell center.

If we define a frequency reuse efficiency, η_f, as in Equation 6.14a, then E_b/I_0 is given by Equation 6.15.

$$\eta_f = \frac{1}{[1 + 6 \times k_1 + 12 \times k_2 + 18 \times k_3 + \cdots]} \tag{6.14a}$$

$$\frac{S}{I} = \frac{\eta_f}{(N_u - 1)} \tag{6.14b}$$

$$\frac{E_b}{I_0} = \frac{B_w}{R} \times \frac{\eta_f}{(N_u - 1)} \tag{6.15}$$

This equation does not include the effect of background thermal and spurious noise (i.e., ρ) in the spreading bandwidth B_w. Including this as an additive degradation term in the denominator results in a bit energy-to-interference ratio of:

$$\frac{E_b}{I_0} = \frac{B_w}{R} \times \frac{\eta_f}{(N_u - 1) + \rho/S} \tag{6.16}$$

Note that from Equation 6.16 the capacity of the DS-CDMA system is reduced by ρ/S which is the ratio of background thermal plus spurious noise to power level.

For a fixed $G_p = B_w/R$, one way to increase the capacity of the DS-CDMA system is to reduce the required E_b/I_0, which depends upon the modulation and coding scheme. By using a powerful coding scheme, the E_b/I_0 ratio can be reduced, but this increases system complexity. Also, it is not possible to reduce the E_b/I_0, ratio indefinitely. The only other way to increase the system capacity is to reduce the interference. Two approaches are used: one is based on the natural behavior of human speech and the other is based on the application of the sectorized antennas. From experimental studies it has been found that typically in a full duplex 2-way voice conversation, the duty cycle of each voice is, on the average, less than 40%. Thus, for the remaining period of time the interference induced by the speaker can be eliminated. Since the channel is shared among all the users, noise induced in the desired channel is reduced due to the silent interval of other interfering channels. It is not cost-effective to exploit the voice activity in the FDMA or TDMA system because of the time delay associated with reassigning the channel resource during the speech pauses. If we define v_f as the voice activity factor (<1), then Equation 6.16 can be written as:

$$\frac{E_b}{I_0} = \frac{\eta_f}{v_f} \times \frac{B_w}{R} \times \frac{1}{(N_u - 1) + \rho/S} \tag{6.17a}$$

$$(N_u - 1) + \frac{\rho}{S} = \left[\frac{\eta_f}{v_f}\right] \times \left[\frac{B_w}{R}\right] \times \left[\frac{I_0}{E_b}\right] \tag{6.17b}$$

The equation to determine the capacity of a DS-CDMA system should also include additional parameters to reflect the bandwidth efficiency factor, the capacity degradation factor due to imperfect power control, and the number of sectors in the cell-site antenna. Equation 6.17b is augmented by these additional factors to provide the following equation for DS-CDMA capacity at one cell:

$$N_u = \frac{\eta_f \eta_b c_d \lambda}{v_f} \times \frac{B_w}{R \times (E_b/I_0)} + 1 - \frac{\rho}{S} \tag{6.18a}$$

Equation 6.18a can be rewritten as Equation 6.18b by neglecting the last two terms.

$$N_u = \frac{\eta_f \eta_b c_d \lambda}{v_f} \times \frac{B_w}{R \times (E_b/I_0)} \tag{6.18b}$$

where:

η_f = frequency reuse efficiency <1
η_b = bandwidth efficiency factor <1
c_d = capacity degradation factor to account for imperfect APC <1
v_f = voice activity factor <1
B_w = one-way bandwidth of the system
R = information bit rate plus overhead
E_b = energy per bit of the desired signal
E_b/I_0 = desired energy-to-interference ratio (dependent on quality of service)
λ = efficiency of sector-antenna in cell (< G, number of sectors in the cell-site antenna)

For digital voice transmission, E_b/I_0 is the required value for a bit error rate (BER) of about 10^{-3} or better, and η_f depends on the quality of the diversity. Under the most optimistic assumption, $\eta_f < 0.5$. The voice activity factor, v_f is usually assumed to be less than or equal to 0.6. E_b/I_0 for a BER of 10^{-3} can be as high as 63 (18 dB) if no coding is used and as low as 5 (7 dB) for a system using a powerful coding scheme. The capacity degradation factor, c_d will depend on the implementation but will always be less than 1.

Example 6.6

Calculate the capacity and spectral efficiency of the DS-CDMA system with an omnidirectional cell using the following data:

- bandwidth efficiency $\eta_b = 0.9$
- frequency reuse efficiency $\eta_f = 0.45$

- capacity degradation factor $c_d = 0.8$
- voice activity factor $v_f = 0.4$
- information bit rate $R = 16.2\,\text{kbps}$
- $E_b/I_0 = 7\,\text{dB}$
- one-way system bandwidth $B_w = 12.5\,\text{MHz}$

Neglect other sources of interference.

Solution

$E_b/I_0 = 5.02\ (7\,\text{dB})$

$$N_u = \frac{0.45 \times 0.9 \times 0.8 \times 1}{0.4} \times \frac{12.5 \times 10^6}{16.2 \times 10^3 \times 5.02}$$

$N_u = 124.5$ (say 125)

The spectral efficiency, $\eta = \dfrac{125 \times 16.2}{12.5 \times 10^3} = 0.162\,\text{bits/sec/Hz}$

In these calculations, an omnidirectional antenna is assumed. If a three sector antenna (i.e., $G = 3$) is used at a cell site with $\lambda = 2.6$, the capacity will be increased to 325 mobile users per cell, and spectral efficiency will be 0.421 bits/sec/Hz.

6.7 Comparison of DS-CDMA vs. TDMA System Capacity

Using Equations 6.7 and 6.18b with $v_f = 1$ (no voice activity) for TDMA and $\lambda = 1.0$ (omnidirectional cell) for DS-CDMA the ratio of the cell capacity for the DS-CDMA and TDMA systems is given as:

$$\frac{N_{\mathrm{CDMA}}}{N_{\mathrm{TDMA}}} = \frac{c_d N \eta_f}{E_b/I_0} \times \frac{1}{v_{f_{\mathrm{cdma}}}} \times \frac{1}{\mu} \times \frac{R_{\mathrm{TDMA}}}{R_{\mathrm{CDMA}}} \tag{6.19}$$

Example 6.7

Using the data given in Examples 6.4 and 6.6, compare the capacity of the DS-CDMA and TDMA omnidirectional cell.

Solution

$$\frac{N_{\mathrm{CDMA}}}{N_{\mathrm{TDMA}}} = \frac{0.8 \times 19 \times 0.45}{5.02} \times \frac{1}{0.4} \times \frac{1}{2} \times \frac{16.2}{16.2} = 1.703$$

6.8 Frequency Hopping Spread Spectrum with M-*ary* Frequency Shift Keying

The FHSS system uses M-*ary* frequency shift keying modulation (MFSK) and involves the hopping of the carrier frequency in a random manner. It uses MFSK, in which $b = \log_2 M$ information bits determine which one of M frequencies is to be used [19]. The portion of the M-*ary* signal set is shifted pseudo-randomly by the frequency synthesizer over a hopping bandwidth, B_{ss}. A typical block diagram is shown in Figure 6.8.

In a conventional MFSK system, the data symbol is modulated on a carrier whose frequency is pseudo-randomly determined. The frequency synthesizer produces a transmission tone based on simultaneous dictates of the pseudo-noise (PN) code (see Chapter 11) and the data. At each frequency hop time a PN generator feeds the frequency synthesizer a frequency word (a sequence of L chips), which dictates one of $2L$ symbol-set positions. The FH bandwidth, B_{ss}, and the minimum frequency spacing between consecutive hop positions, Δf, dictate the minimum number of chips required in the frequency word.

Example 6.8

A hopping bandwidth, B_{ss}, of 600 MHz and a frequency step size, Δf, of 400 Hz are used. What is the minimum number of PN chips that are required for each frequency word?

Solution

$$\text{Number of tones contained in } B_{ss} = \frac{B_{ss}}{\Delta f} = \frac{600 \times 10^6}{400} = 1.5 \times 10^6$$

$$\text{Minimum number of chips required} = \left\lceil \log_2(1.5 \times 10^6) \right\rceil = 20\,\text{chips}$$

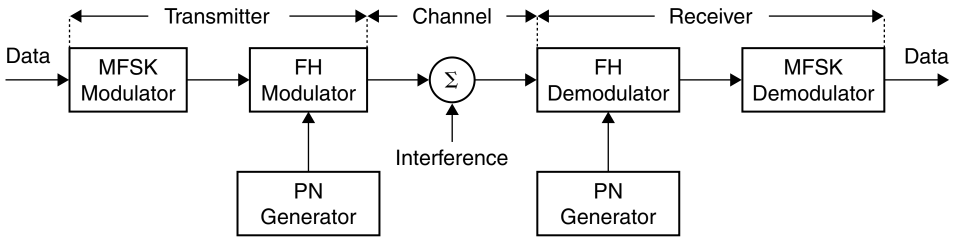

Figure 6.8 Frequency hopping using MFSK.

6.9 Orthogonal Frequency Division Multiplexing (OFDM)

In this section we briefly introduce OFDM. For more details readers should refer to [19]. OFDM uses three transmission principles, multirate, multisymbol, and multicarrier. OFDM is similar to frequency division multiplexing (FDM). OFDM distributes the data over a large number of carriers that are spaced apart at precise frequencies. The spacing provides the orthogonality in this technique, which prevents the demodulator from seeing frequencies other than their own.

Multiple Input, Multiple Output-OFDM (MIMO-OFDM) uses multiple antennas to transmit and receive radio signals. MIMO-OFDM allows service providers to deploy a broadband wireless access system that has non-line-of-sight (NLOS) functionality. MIMO-OFDM takes advantage of the multipath properties of the environment using base station antennas that do not have LOS. The MIMO-OFDM system uses multiple antennas to simultaneously transmit data in small pieces to the receiver, which can process the data flow and put it back together. This process, called *spatial multiplexing*, proportionally boosts the data transmission speed by a factor equal to the number of transmitting antennas. In addition, since all data is transmitted both in the same frequency band and with separate spatial signatures, this technique utilizes spectrum efficiently. VOFDM (vector OFDM) uses the concept of MIMO technology.

We consider a data stream operating at R bps and an available bandwidth of $N\Delta f$ centered at f_c. The entire bandwidth could be used to transmit a data stream, in which case the bit duration would be $1/R$. By splitting the data stream into N substreams using a serial-to-parallel converter, each substream has a data rate of R/N and is transmitted on a separate subcarrier, with spacing between adjacent subcarriers of Δf (see Figure 6.9). The bit duration is N/R. The advantage of OFDM is that on a multiple channel the multipath is reduced relative to the symbol interval by a ratio of $1/N$ and thus imposes less distortion in each modulated symbol. OFDM overcomes inter-symbol interference (ISI) in a multipath environment. ISI has a greater impact at higher data rates because the distance between bits or symbols is smaller. With OFDM, the data rate is reduced by a factor of N, which increases the symbol duration by a factor of N. Thus, if the symbol duration is T_s for the source stream, the duration of OFDM signals is NT_s. This significantly reduces the effect of ISI. As a design criterion, N is selected so that NT_s is significantly greater than τ_{rms} (rms delay spread) of the channel. With the use of OFDM, it may not be necessary to deploy an equalizer. OFDM is an ideal solution for broadband communications, because increasing the data rate is simply a matter of increasing the number of subcarriers. To avoid overlap between consecutive symbols, a time guard is enforced between the transmissions of two OFDM pulses that will reduce the effective data rate. Also, some subcarriers are devoted to synchronization of signal, and some are reserved for redundancy.

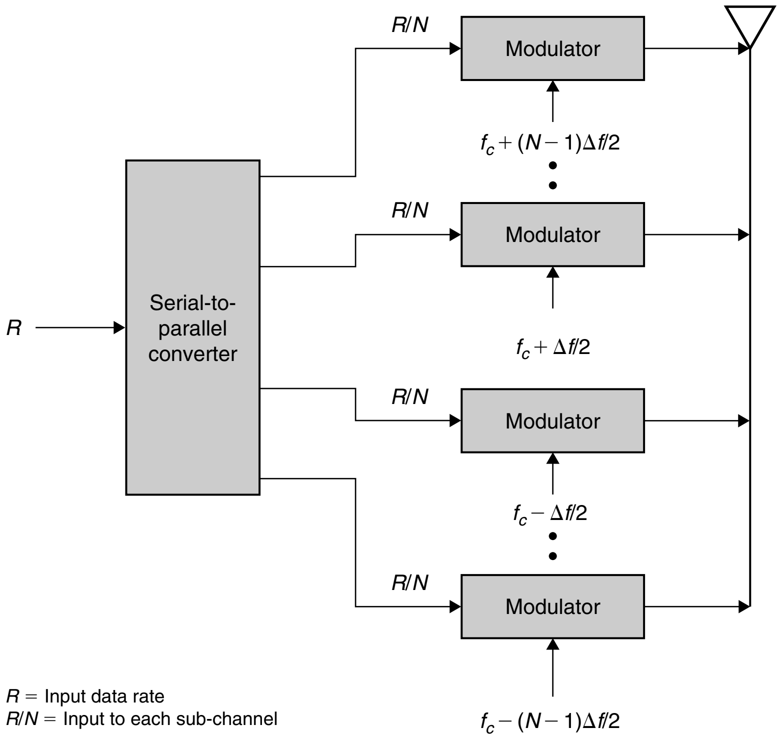

Figure 6.9 Orthogonal frequency division multiplexing (OFDM).

The most important feature of OFDM is the orthogonal relationship between the subcarrier signals. Orthogonality allows the OFDM subcarriers to overlap each other without interference. OFDM uses FH to create a spread spectrum system. FH has several advantages over DSSS, for example, no near-far problem, easier synchronization, less complex receivers, and so on.

In the OFDM the input information sequence is first converted into parallel data sequences and each serial/parallel converter output is multiplied with spreading code. Data from all subcarriers is modulated in baseband by inverse fast Fourier transform (IFFT) and converted back into serial data. The guard interval is inserted between symbols to avoid ISI caused by multipath fading and finally the signal is transmitted after RF up-conversion. At the receiver, after down-conversion, the m-subcarrier component corresponding to the received data is first coherently detected with FFT and then multiplied with gain to combine the energy of the received signal scattered in the frequency domain (see Figure 6.10).

Wireless Local Area Networks (WLAN) development is ongoing for wireless point-to-point and point-to-multipoint configurations using OFDM technology.

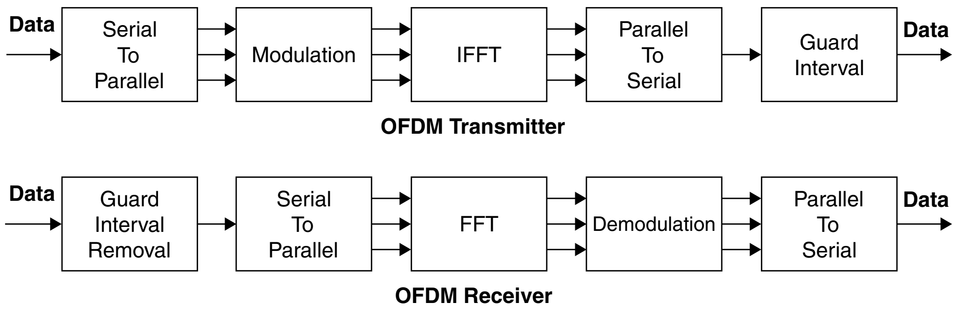

Figure 6.10 IEEE 802.11 a Transmit and Receive OFDM.

In a supplement to the IEEE 802.11 standard, the IEEE 802.11 working group published IEEE 802.11a, which outlines the use of OFDM in the 5.8 GHz band.

The basic principal of operation is to divide a high-speed binary signal to be transmitted into a number of lower data rate subcarriers. There are 48 data subcarriers and 4 pilot subcarriers for a total of 52 subcarriers. Each lower data rate bit stream is used to modulate a separate subcarrier from one of the channels in the 5 GHz band. Prior to transmission the data is encoded using convolutional code (see Chapter 8) of rate, $R = 1/2$ and bit interleaved for the desired data rate. Each bit is then mapped into a complex number according to the modulation type and subdivided in 48 data subcarriers and 4 pilot subcarriers. The subcarriers are combined using an IFFT and transmitted. At the receiver, the carrier is converted back to a multicarrier lower data rate form using FFT. The lower data subcarriers are combined to form a high rate data unit.

6.10 Multicarrier DS-CDMA (MC-DS-CDMA)

Future wireless systems such as a fourth-generation (4G) system will need flexibility to provide subscribers with a variety of services such as voice, data, images, and video. Because these services have widely differing data rates and traffic profiles, future generation systems will have to accommodate a wide variety of data rates. DS-CDMA has proven very successful for large-scale cellular voice systems, but there are concerns whether DS-CDMA will be well-suited to non-voice traffic. The DS-CDMA system suffers inter-symbol interference (ISI) and multi-user interference (MUI) caused by multipath propagation, leading to a high loss of performance.

With OFDM, the time dispersive channel is seen in the frequency domain as a set of parallel independent flat subchannels and can be equalized at a low complexity. There are potential benefits to combining OFDM and DS-CDMA. Basically the frequency-selective channel is first equalized in the frequency domain using the

OFDM modulation technique. DS-CDMA is applied on top of the equalized channel, keeping the orthogonality properties of spreading codes. The combination of OFDM and DS-CDMA is used in MC-DS-CDMA. MC-DS-CDMA [4,5,12,25] marries the best of the OFDM and DS-CDMA world and, consequently, it can ensure good performance in severe multipath conditions. MC-DS-CDMA can achieve very large average throughput. To further enhance the spectral efficiency of the system, some form of adaptive modulation can be used.

Basically, three main designs exist in the literature, namely, MC-CDMA, MC-DS-CDMA, and multitone (MT)-CDMA. In MC-CDMA, the spreading code is applied across a number of orthogonal subcarriers in the frequency domain. In MC-DS-CDMA, the data stream is first divided into a number of substreams. Each substream is spread in time through a spreading code and then transmitted over one of a set of orthogonal subcarriers. In MT-CDMA the system undergoes similar operations as MC-DS-CDMA except that the different subcarriers are not orthogonal after spreading. This allows higher spectral efficiencies and longer spreading codes; however, different substreams interfere with one other. The MC-DS-CDMA transmitter spreads the original data stream over different orthogonal subcarriers using a given spreading code in the frequency domain.

6.11 Random Access Methods

So far we have discussed the reservation-based schemes, now we focus on random-access schemes [8]. When each user has a steady flow of information to transmit (for example, a data file transfer or a facsimile transmission), reservation-based access methods are useful as they make an efficient use of communication resources. However, when the information to be transmitted is bursty in nature, the reservation-based access methods result in wasting communication resources. Furthermore, in a cellular system where subscribers are charged based on a channel connection time, the reservation-based access methods may be too expensive to transmit short messages. Random-access protocols provide flexible and efficient methods for managing a channel access to transmit short messages. The random-access methods give freedom for each user to gain access to the network whenever the user has information to send. Because of this freedom, these schemes result in contention among users accessing the network. Contention may cause collisions and may require retransmission of the information. The commonly used random-access protocols are pure ALOHA, slotted-ALOHA, and CSMA/CD. In the following section we briefly describe details of each of these protocols and provide the necessary throughput expressions.

6.11.1 Pure ALOHA

In the pure ALOHA [18,23] scheme, each user transmits information whenever the user has information to send. A user sends information in packets. After sending a packet, the user waits a length of time equal to the round-trip delay for an acknowledgment (ACK) of the packet from the receiver. If no ACK is received, the packet is assumed to be lost in a collision and it is retransmitted with a randomly selected delay to avoid repeated collisions.* The normalized throughput S (average new packet arrival rate divided by the maximum packet throughput) of the pure ALOHA protocol is given as:

$$S = Ge^{-2G} \tag{6.20}$$

where G = normalized offered traffic load

From Equation 6.20 it should be noted that the maximum throughput occurs at traffic load $G = 50\%$ and is $S = 1/2e$. This is about 0.184. Thus, the best channel utilization with the pure ALOHA protocol is only 18.4%.

6.11.2 Slotted ALOHA

In the slotted-ALOHA [23] system, the transmission time is divided into time slots. Each time slot is made exactly equal to packet transmission time. Users are synchronized to the time slots, so that whenever a user has a packet to send, the packet is held and transmitted in the next time slot. With the synchronized time slots scheme, the interval of a possible collision for any packet is reduced to one packet time from two packet times, as in the pure ALOHA scheme. The normalized throughput S for the slotted-ALOHA protocol is given as:

$$S = Ge^{-G} \tag{6.21}$$

where G = normalized offered traffic load

The maximum throughput for the slotted ALOHA occurs at $G = 1.0$ (Equation 6.21) and it is equal to $1/e$ or about 0.368. This implies that at the maximum throughput, 36.8% of the time slots carry successfully transmitted packets. The best channel utilization with the slotted ALOHA protocol is 36.8%—twice the pure ALOHA protocol.

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*It should be noted that the protocol on CDMA access channels as implemented in TIA IS-95-A is based upon the pure ALOHA approach. The mobile station randomizes its attempt for sending a message on the access channel and may retry if an acknowledgment is not received from the base station. For further details, one should reference Section 6.6.3.1.1.1 of TIA IS-95-A.

6.11.3 Carrier Sense Multiple Access (CSMA)

The carrier sense multiple access (CSMA) [8,18] protocols have been widely used in both wired and wireless LANs. These protocols provide enhancements over the pure and slotted ALOHA protocols. The enhancements are achieved through the use of the additional capability at each user station to sense the transmissions of other user stations. The carrier sense information is used to minimize the length of collision intervals. For carrier sensing to be effective, propagation delays must be less than packet transmission times. Two general classes of CSMA protocols are nonpersistent and p-persistent.

- **Nonpersistent CSMA:** A user station does not sense the channel continuously while it is busy. Instead, after sensing the busy condition, it waits for a randomly selected interval of time before sensing again. The algorithm works as follows: if the channel is found to be idle, the packet is transmitted; or if the channel is sensed busy, the user station backs off to reschedule the packet to a later time. After backing off, the channel is sensed again, and the algorithm is repeated again.
- **p-persistent CSMA:** The slot length is typically selected to be the maximum propagation delay. When a station has information to transmit, it senses the channel. If the channel is found to be idle, it transmits with probability p. With probability q = 1 − p, the user station postpones its action to the next slot, where it senses the channel again. If that slot is idle, the station transmits with probability p or postpones again with probability q. The procedure is repeated until either the frame has been transmitted or the channel is found to be busy. If the station initially senses the channel to be busy, it simply waits one slot and applies the above procedure.
- **1-persistent CSMA:** 1-persistent CSMA is the simplest form of the p-persistent CSMA. It signifies the transmission strategy, which is to transmit with probability 1 as soon as the channel becomes idle. After sending the packet, the user station waits for an ACK, and if it is not received within a specified amount of time, the user station waits for a random amount of time, and then resumes listening to the channel. When the channel is again found to be idle, the packet is retransmitted immediately.

For more details, the reader should refer to [18].

The throughput expressions for the CSMA protocols are:

- **Unslotted nonpersistent CSMA**

$$S = \frac{Ge^{-aG}}{G(1 + 2a) + e^{-aG}} \tag{6.22}$$

- **Slotted nonpersistent CSMA**

$$S = \frac{aGe^{-aG}}{1 - e^{-aG} + a} \tag{6.23}$$

- **Unslotted 1-persistent CSMA**

$$S = \frac{G[1 + G + aG(1 + G + (aG)/2)]e^{-G(1 + 2a)}}{G(1 + 2a) - (1 - e^{-aG}) + (1 + aG)e^{-G(1 + a)}} \tag{6.24}$$

- **Slotted 1-persistent CSMA**

$$S = \frac{Ge^{-G(1 + a)}[1 + a - e^{-aG}]}{(1 + a)(1 - e^{-aG}) + ae^{-G(1 + a)}} \tag{6.25}$$

where:

S = normalized throughput
G = normalized offered traffic load
a = τ / T_p
τ = maximum propagation delay
T_p = packet transmission time

Example 6.9

We consider a WLAN installation in which the maximum propagation delay is 0.4 sec. The WLAN operates at a data rate of 10 Mbps, and packets have 400 bits. Calculate the normalized throughput with: (1) an unslotted nonpersistent, (2) a slotted persistent, and (3) a slotted 1-persistent CSMA protocol.

Solution

$$T_p = \frac{400}{10} = 40\,\mu\text{s}$$

$$a = \frac{\tau}{T_p} = \frac{0.4}{40} = 0.01$$

$$G = \frac{40 \times 10^{-6} \times 10 \times 10^{6}}{400} = 1$$

- Slotted nonpersistent:

$$S = \frac{0.01 \times 1 \times e^{-0.01}}{1 - e^{-0.01} + 0.01} = 0.495$$

- Unslotted nonpersistent:

$$S = \frac{1 \times e^{-0.01}}{(1 + 0.02) + e^{-0.01}} = 0.493$$

- Slotted 1-persistent:

$$S = \frac{e^{-1.01}(1 + 0.01 - e^{-0.01})}{(1 + 0.01)(1 - e^{-0.01}) + 0.01e^{-1.01}} = 0.531$$

6.11.4 Carrier Sense Multiple Access with Collision Detection

A considerable performance improvement in the basic CSMA protocols can be achieved by means of the *carrier sense multiple access with collision detection* (CSMA/CD) technique. The CSMA/CD protocols are essentially the same as those for CSMA with addition of the collision-detection feature. Similar to CSMA protocols, there are nonpersistent, 1-persistent, and p-persistent CSMA/CD protocols. More details about CSMA/CD protocols can be found in [27].

When a CSMA/CD station senses that a collision has occurred, it immediately stops transmitting its packets and sends a brief jamming signal to notify all stations of this collision. Collisions are detected by monitoring the analog waveform directly from the channel. When signals from two or more stations are present simultaneously, the composite waveform is distorted from that of a single station. This is manifested in the form of larger than normal voltage amplitude on the cable. In the Ethernet the collision is recognized by the transmitting station, which goes into a retransmission phase based on an exponential random backoff algorithm.

The normalized throughput for unslotted nonpersistent and slotted nonpersistent CSMA/CD is given as:

Unslotted nonpersistent CSMA/CD

$$S = \frac{Ge^{-aG}}{Ge^{-aG} + bG(1 - e^{-aG}) + 2aG(1 - e^{-aG}) + (2 - e^{-aG})} \tag{6.26}$$

where b = jamming signal length

Slotted nonpersistent CSMA/CD

$$S = \frac{aGe^{-aG}}{aGe^{-aG} + b(1 - e^{-aG} - aGe^{-aG}) + a(2 - e^{-aG} - aGe^{-aG})} \tag{6.27}$$

While these collision detection mechanisms are a good idea on a wired local area network (LAN), they cannot be used on a wireless local area network (WLAN) environment for two main reasons:

- Implementing a collision detection mechanism would require the implementation of a full duplex radio capable of transmitting and receiving at the same time—an approach that would increase the cost significantly.
- In a wireless environment we cannot assume that all stations hear each other (which is the basic assumption of the collision detection scheme), and the fact that a station wants to transmit and senses the medium as free does not necessarily mean that the medium is free around the receiver area.

6.11.5 Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA)

IEEE 802.11 uses a protocol known as *carrier sense multiple access with collision avoidance* (CSMA/CA) or distributed coordination function (DCF). CSMA/CA attempts to avoid collisions by using *explicit packet acknowledgment* (ACK), which means an ACK packet is sent by the receiving station to confirm that the data packet arrived intact.

The CSMA/CA protocol works as follows. A station wishing to transmit senses the medium, if the medium is busy (i.e., some other station is transmitting) then the station defers its transmission to a later time. If no activity is detected, the station waits an additional, randomly selected period of time and then transmits if the medium is still free. If the packet is received intact, the receiving station issues an ACK frame that, once successfully received by the sender, completes the process. If the ACK frame is not detected by the sending station, either because the original data packet was not received intact or the ACK was not received intact, a collision is assumed to have occurred and the data packet is transmitted again after waiting another random amount of time. The CSMA/CA provides a way to share access over the medium. This explicit ACK mechanism also handles interference and other radio-related problems very effectively. However, it does add some overhead to 802.11 that 802.3 does not have, so that an 802.11 WLAN will always have slower performance than the equivalent Ethernet LAN (802.3).

The CSMA/CA protocol is very effective when the medium is not heavily loaded since it allows stations to transmit with minimum delay. But there is always a chance of stations simultaneously sensing the medium as being free and transmitting at the same time, causing a collision. These collisions must be identified, so that the media access control (MAC) layer can retransmit the packet by itself and not by the upper layers, which would cause significant delay. In particular, the hidden node and exposed node problems should be addressed by MAC. Both of them give rise to many performance problems including throughput degradtion, unfair throughput distribution, and throughput instability (see Chapter 18 for details).

The IEEE 802.11 uses a collision avoidance (CA) mechanism together with a positive ACK. The MAC layer of a station wishing to transmit senses the medium. If the medium is free for a specified time (called *distributed inter-frame space* (DIFS)), then the station is able to transmit the packet; if the medium is busy (or becomes busy during the DIFS interval) the station defers using the *exponential backoff algorithm*.

This scheme implies that, except in cases of very high network congestion, no packets will be lost, because retransmission occurs each time a packet is not acknowledged. This entails that all packets sent will reach their destination in sequence.

The IEEE 802.11 MAC layer provides cyclic redundancy check (*CRC*) *checksum* and *packet fragmentation*. Each packet has a CRC checksum calculated and attached to ensure that the data was not corrupted in transmit. Packet fragmentation is used to segment large packets into smaller units when sent over the medium. This is useful in very congested environments or when interference is a factor, since large packets have a better chance of being corrupted. This technique reduces the need for retransmission in many cases and improves overall wireless network performance. The MAC layer is responsible for reassembling fragments received, rendering the process transparent to higher-level protocols. The following are some of the reasons it is preferable to use smaller packets in a WLAN environment.

- Due to higher BER of a radio link, the probability of a packet getting corrupted increases with packet size.
- In case of packet corruption (either due to collision or interference), the smaller the packet, the less overhead it needs to retransmit.

A simple *stop-and-wait* algorithm is used at the MAC sublayer. In this mechanism the transmitting station is not allowed to transmit a new fragment until one of the following happens:

- Receives an ACK for the fragment, or
- Decides that the fragment was retransmitted too many times and drops the whole frame.

Exponential backoff scheme is used to resolve contention problems among different stations wishing to transmit data at the same time. When a station goes into the backoff state, it waits an additional, randomly selected number of time slots known as a contention window (in 802.11b a slot has a 20 μs duration and the random number must be greater than 0 and smaller than a maximum value referred to as a contention window (CW)). During the wait, the station continues sensing the medium to check whether it remains free or if another transmission begins. At the end of its window, if the medium is still free the station can send its frame. If during the window another station begins transmitting data, the backoff counter is frozen and counting down starts again as the channel returns to the idle state.

There is a problem related to the CW dimension. With a small CW, if many stations attempt to transmit data at the same time it is possible that some of them may have the same backoff interval. This means that there will continuously be collisions, with serious effects on network performance. On the other hand, with a large CW, if a few stations wish to transmit data they will likely have long backoff delays resulting in degradation of network performance. The solution is to use an exponentially growing CW size. It starts from a small value ($CW_{min} = 31$) and doubles after each collision, until it reaches the maximum value CW_{max} ($CW_{max} = 1023$). In 802.11 the backoff algorithm is executed in three cases:

1. When the station senses the medium busy before the first transmission of a packet
2. Before each retransmission
3. After a successful transmission

This is necessary to avoid a single host wanting to transmit a large quantity of data and occupying the channel for too long, denying access to all other stations. The backoff mechanism is not used when the station decides to transmit

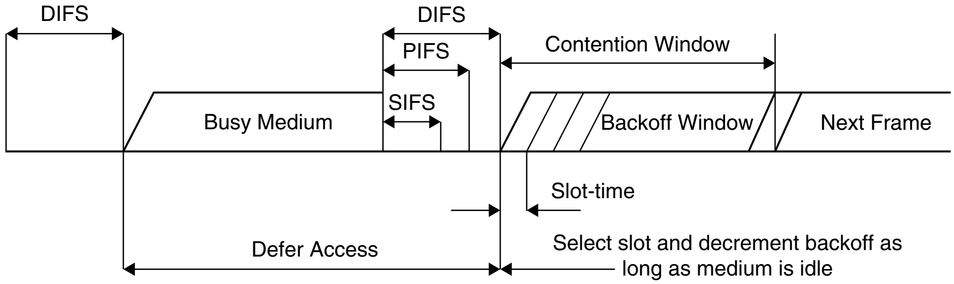

Figure 6.11 CSMA/CA in IEEE 802.11b.

a new packet after an idle period and the medium has been free for more than a distributed inter-frame space (DIFS) (see Figure 6.11).

To support time-bounded services, the IEEE 802.11 standard defines the point coordinate function (PCF) to let stations have priority access to the wireless medium, coordinated by a station called point coordinate (PC). The PCF has higher priority than DIFS, because it may start transmissions after a shorter duration than DIFS; this time space is called PCF inter frame space (PIFS), which is 25 μs for IEEE 802.11 and larger than SIFS.

$$\text{The transmission time for a data frame} = \left(\text{PLCP} + \frac{D}{R}\right)\ \mu\text{s}$$

where:

PLCP = the time required to transmit the physical layer convergence protocol (PLCP)
D = the frame size
R = the channel bit rate

$$\text{CSMA/CA packet transmission time} = \text{BO} + \text{DIFS} + 2\text{PLCP} + \frac{D}{R} + \text{SIFS} + \frac{A}{R}\ \mu\text{s}$$

where:

A = the ACK frame size
BO = the backoff time
DIFS = the distributed inter-frame space
SIFS = the short inter-frame space
PIFS = the point coordinate interframe space

6.12 Idle Signal Casting Multiple Access

In the CSMA scheme, each terminal must be able to detect the transmissions of all other terminals. However, not all packets transmitted from different terminals can be sensed, or terminals may be hidden from each other by buildings or some other obstacles. This is known as the *hidden terminal problem*, which severely degrades the throughput of the CSMA. The idle signal casting multiple access (ISMA) system transmits an idle/busy signal from the base station to indicate the presence or absence of another terminal's transmission. The ISMA and CSMA are basically the same. In the CSMA, each terminal must listen to all other terminals, whereas in the ISMA, each terminal is informed from the base station of the other terminals' transmission. Similar to CSMAs, there are nonpersistent ISMAs and 1-persistent ISMAs.

6.13 Packet Reservation Multiple Access

Packet reservation multiple access (PRMA) allows a variety of information sources to share the same communication channel and obtains a statistical multiplexing effect. In PRMA, time is divided into frames, each of which consists of a fixed number of time slots. For voice terminals, voice activity detection is adopted. The voice signal comprises a sequence of talk spurts. At the beginning of a talk spurt, the terminal transmits the first packet based on slotted ALOHA. Once the packet is transmitted successfully, that terminal is allowed to use the same time slots in the succeeding frames (reservation is made). The reservation is kept until the end of the talk spurt. The status "reserved" or "unreserved" of each slot is broadcast from the base station.

6.14 Error Control Schemes for Link Layer

Error control schemes for the link layer are used to improve the performance of mobile communication systems [8]. Several automatic repeat request (ARQ) schemes are used. At the physical layer of wireless mobile communication systems, error detection and correction techniques such as forward error correction (FEC) schemes are used. For some of the data services, higher layer protocols use ARQ schemes to enable retransmission of any data frames in which an error is detected. The ARQ schemes are classified as follows [23]:

- **Stop and Wait:** The sender transmits the first packet numbered 0 after storing a copy of that packet. The sender then waits for an ACK numbered 0, (ACK0) of that packet. If the ACK0 does not arrive before a time-out, the sender transmits another copy of the first packet. If the ACK0 arrives before a time-out, the sender discards the copy of the first packet and is ready to transmit the next packet, which it numbers 1. The sender repeats the previous steps, with numbers 0 and 1 interchanged. The advantages of the Stop and Wait protocol are its simplicity and its small buffer requirements. The sender needs to keep only the copy of the packet that it last transmitted, and the receiver does not need to buffer packets at the data link layer. The main disadvantage of the Stop and Wait protocol is that it does not use the communication link very efficiently.

The total time taken to transmit a packet and to prepare for transmitting the next one is

$$T = T_p + 2T_{\text{prop}} + 2T_{\text{proc}} + T_a \tag{6.28}$$

where:

T = total time for transmission time
T_p = transmission time for a packet
T_{prop} = propagation time of a packet or an ACK
T_{proc} = processing time for a packet or an ACK
T_a = transmission time for an ACK

The protocol efficiency without any error is:

$$\eta(0) = \frac{T_p}{T} \tag{6.29}$$

If p is the probability that a packet or its ACK is corrupted by transmission errors, and a successful transmission of a packet and its ACK takes T seconds and occurs with probability $1 - p$, the protocol efficiency for full duplex (FD) is given as:

$$\eta_{\mathrm{FD}} = \frac{(1-p)T_p}{(1-p)T + pT_p} \tag{6.30}$$

- **Selective Repeat Protocol (SRP):** In case of the SRP, only the selected packets are retransmitted. The data link layer in the receiver delivers exactly one copy of every packet in the correct order. The data link layer in the receiver may get the packets in the wrong order from the physical layer. This occurs, for example, when transmission errors corrupt the first packet and not the second one. The second packet arrives correctly at the receiver before the first. The data link layer in the receiver uses a buffer to store the packets that arrive out of order. Once the data link layer in the receiver has a consecutive group of packets in its buffer, it can deliver them to the network layer. The sender also uses a buffer to store copies of the unacknowledged packets. The number of the packets which can be held in the sender/receiver buffer is a design parameter.

Let W = the number of packets which the sender and receiver buffers can each hold and SRP = number of packets in modulo 2W. The protocol efficiency without any error and with a packet error probability of p is given as:

$$\eta(0) = \min\left\{\frac{WT_p}{T}, 1\right\} \tag{6.31}$$

For very large W, the protocol efficiency is

$$\eta(p) = 1 - p \tag{6.32}$$

where:

WT_p = time-out

$$\eta(p) = \frac{2 + p(W-1)}{2 + p(3W-1)} \tag{6.33}$$

SRP is very efficient, but it requires buffering packets at both the sender and the receiver.

- **Go-Back-N (GBN):** The Go-Back-N protocol allows the sender to have multiple unacknowledged packets without the receiver having to store packets. This is done by not allowing the receiver to accept packets that are out of order. When a time-out timer expires for a packet, the transmitter resends that packet and all subsequent packets. The Go-Back-N protocol improves on the efficiency of the Stop and Wait protocol, but is less efficient than SRP. The protocol efficiency for full duplex is given as:

$$\eta_{FD} = \frac{1}{1 + \left(\frac{p}{1-p}\right)W} \tag{6.34}$$

- **Window-control Operation Based on Reception Memory (WORM) ARQ:** In digital cellular systems, bursty errors occur by multipath fading, shadowing, and handoffs. The typical bit-error rate fluctuates from 10^{-1} to 10^{-6}. Therefore, the conventional ARQ schemes do not operate well in a digital cellular system. WORM ARQ has been suggested for control of dynamic error characteristics. It is a hybrid scheme that combines SRP GBN protocol. GBN protocol is chosen in the severe error condition whereas SRP is selected in the normal error condition.
- **Variable Window and Frame Size GBN and SRP [24]:** Since wireless systems have bursty error characteristics, the error control schemes should have a dynamic adaptation to a bursty channel environment. The SRP and GBN with variable window and frame size have been proposed to improve error control in wireless systems. Table 6.2 provides the window and frame size for different BER. If the error rate increases, the window and frame size are decreased. In the case of the error rate being small, the window and frame size are increased. The optimum threshold values of BER, window and frame size were obtained through computer simulation.

Table 6.2 Bit-error rate versus window and size.

Bit-error rate (BER)	Window size (W)	Frame size (bits)
BER $\le 10^{-4}$	32	172
$10^{-4} <$ BER $< 10^{-3}$	8	80
$10^{-3} <$ BER $< 10^{-2}$	4	40
$10^{-2} <$ BER	2	16

Example 6.10

We consider a WLAN in which the maximum propagation delay is 4 sec. The WLAN operates at a data rate of 10 Mbps. The data and ACK packets are of 400 and 20 bits, respectively. The processing time for a data or ACK packet is 1 sec. If the probability p that a data packet or its ACK can be corrupted during transmission is 0.01, find the data link protocol efficiency with (1) Stop and Wait protocol—full duplex, (2) SRP with window size W = 8, and (3) Go-Back-N protocol with window size W = 8.

Solution

$$T_p = \frac{400}{10} = 40\mu\text{s}$$

$$T_a = \frac{20}{10} = 2\mu\text{s}$$

$$T_{\text{prop}} = 4\mu\text{s}$$

$$T_{\text{proc}} = 1\mu\text{s}$$

$$T = 40 + 2 \times 4 + 2 \times 1 + 2 = 52\mu\text{s}$$

Stop and Wait:

$$\eta = \frac{(1 - 0.01) \times 40}{(1 - 0.01) \times 52 + 0.01 \times 40} = 0.763$$

SRP:

$$\eta = \frac{2 + 0.01(8 - 1)}{2 + 0.01(24 - 1)} = 0.954$$

GBN:

$$\eta = \frac{1}{1 + 8\left(\frac{0.01}{1 - 0.01}\right)} = 0.925$$

6.15 Summary

The chapter described the access technologies used in wireless communications including reservation-based multiple access and random multiple access. FDMA, TDMA, and CDMA technologies were discussed and their advantages and disadvantages were listed. Illustrated examples were given to show calculations for determining the capacity of TDMA and CDMA systems. Brief descriptions of the FDD, TDD, TDM/TDMA, and TDM/TDMA/FDD approaches were also given. Since packet networks are an important part of wireless networks, we briefly stated the characteristics of the access methods in common use and defined their throughput equations. The common packet protocols such as ALOHA, slotted ALOHA, and Carrier Sense Multiple Access (CSMA) were discussed. We also presented the methods used to control errors for data link protocols.

Problems

6.1 In a proposed TDMA cellular system, the one-way bandwidth of the system is 40 MHz. The channel spacing is 30 kHz and total voice channels in the system are 1333. The frame duration is 40 ms divided equally between six time slots. The system has an individual user data rate of 16.2 kbps in which the speech with error protection has a rate of 13 kbps. Calculate the efficiency of the TDMA system.What is the efficiency of the system with 20, 60, 80 and 100 MHz?

6.2 Recompute the capacity of the GSM system in Example 5.1 when a sectorized system is used. With sectorization, there are 12 channel sets of 39 channels each with three sets assigned at each cell, one for each sector.

6.3 In the IS-54 (TDMA/FDD), the frame duration is 40 ms. The frame contains six time slots. The transmit bit rate is 48.6 kbps. Each time slot carries 260 bits of user information. The total number of 30 kHz voice channels available is 395 and the total system bandwidth is 12.5 MHz. Calculate the access efficiency of the system.

6.4 Calculate the capacity and spectral efficiency (η) of the IS-54 system using the following parameters: $\eta_b = 0.96$, $\mu = 2$ (i.e., $\pi/4$-DQPSK), voice activity factor $v_f = 1.0$, information bit rate = 19.5 kbps, frequency reuse factor = 7 and system bandwidth = 12.5 MHz.

6.5 Calculate the cell capacity and spectral efficiency of a GSM system using the following data: (1) bandwidth efficiency factor = 1, (2) bit efficiency (with GMSK modulation) = 1, (3) voice activity factor = 1, (4) one-way bandwidth of the system = 10 MHz, (4) information bit rate per frame = 270.83 kbps, (5) number of users per frame = 8, and (6) frequency reuse factor = 4.

6.6 Consider a CDMA system that uses QPSK modulation and convolutional coding. The system has a bandwidth of 1.25 MHz and transmits data at 9.6 kbps. Find the number of users that can be supported by the system and bandwidth efficiency. Assume a three-sector antenna system with an effective gain of 2.6, power control efficiency = 90%, and frequency reuse efficiency of 66.67%. A bit-error rate of 10^{-3} is required.

6.7 A QPSK/DSSS WLAN is designed to transmit in the 902- to 928-MHz ISM band. The symbol transmission rate is 0.25 Megasymbols per second. An orthogonal code with eight symbols is used. A bit-error rate of 10^{-5} is required. How many users can be supported by the WLAN? A three-sector antenna with gain = 2.6 is used. Assume frequency reuse efficiency of 66.67% and power control efficiency of 90%.What is the bandwidth efficiency?

6.8 A WLAN accommodates 50 stations running the same application. The transmission rate per station is 2 Mbps and the stations use slotted ALOHA protocol. The total traffic produced by the stations is assumed to form a Poisson process. What is the maximum throughput in Erlangs? What is the maximum throughput in Mbps? What is the maximum throughput in Mbps for each station?

6.9 Consider a WLAN installation in which maximum propagation delay is 0.5 μs. The WLAN operates at a data rate of 12 Mbps, and each packet is 600 bits. Calculate the throughput with (1) an unslotted nonpersistent, (2) a slotted persistent, and (3) a slotted 1-persistent CSMA protocol.

6.10 Consider a WLAN in which the maximum propagation delay is 5 μs. The WLAN operates at a data rate of 12 Mbps. The data and ACK packet are 600 and 24 bits, respectively. The processing time for data or ACK packet is 2 μs. If the probability p that a data packet or its ACK can be corrupted during transmission is 1%, find the data link protocol efficiency with (1) Stop-and-Wait protocol, full duplex, (2) SRP with window size W = 12, and (3) Go-Back-N protocol with window size W = 12.

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