CHAPTER 6

MULTIPLE-ACCESS TECHNIQUES

6.0 INTRODUCTION

Because of the limited amount of bandwidth available, it is necessary to explore methods to allow multiple users to share the available spectrum simultaneously. Three major approaches exist for accomplishing "sharing" of the resources (Coop 1979, Yue 1983, Pahl 1995, Jake 1974, Samp 1997, Stee 1994, Dixo 1994). These are FDMA, TDMA, and CDMA. In frequency division multiple access (FDMA), the available frequency band is divided among different users. In time division multiple access (TDMA), a fixed amount of spectrum is allocated to a number of users, who use the spectrum only for a very short period and share the time on a reserved basis. On the other hand, in code division multiple access (CDMA), the whole bandwidth is shared all the time by all the users, who use different codes (Vite 1979, 1985; Taub 1986; Shap 1994; Scho 1977, 1982). The three forms of multiple access are represented in Figure 6.1.

It is also important that users be allowed to transmit information to the base station and receive information from the base station simultaneously. In other words, systems must also have a full-duplex provision. This may be achieved in the frequency domain or in the time domain. In frequency division duplex (FDD) systems, two distinct bands of frequencies are provided for every user. These bands are separated by a guard band, as shown in Figure 6.2*a*. The forward band will provide the traffic flow from the base station to the mobile unit while the reverse band will provide the traffic flow from the mobile unit to the base station (see details on frequency bands in Section 6.4). For all users and whichever frequency bands they are using, the frequency split between the forward and reverse channels is always the same. Since the transmission and reception are undertaken using the same antenna, a duplexer (Figure 6.2*b*) is needed to separate the two signals. The carrier frequencies of the uplink and downlink should be separated by an amount sufficient to allow the use of low-cost methods to separate the two signals.

In time domain duplex (TDD) systems, time rather than frequency is used to separate the forward and reverse channels, as shown in Figure 6.3. Since the time split between the forward and reverse channels is very small, almost continuous transmission is possible. While a duplexer is required in FDD systems to separate the frequency bands for the forward and reverse channels, no such equipment is required in TDD systems.

The multiple access schemes are used in conjunction with the duplex configurations to provide simultaneous transmission of uplink and downlink information. We therefore have FDMA/FDD, TDMA/FDD, TDMA/TDD, etc.

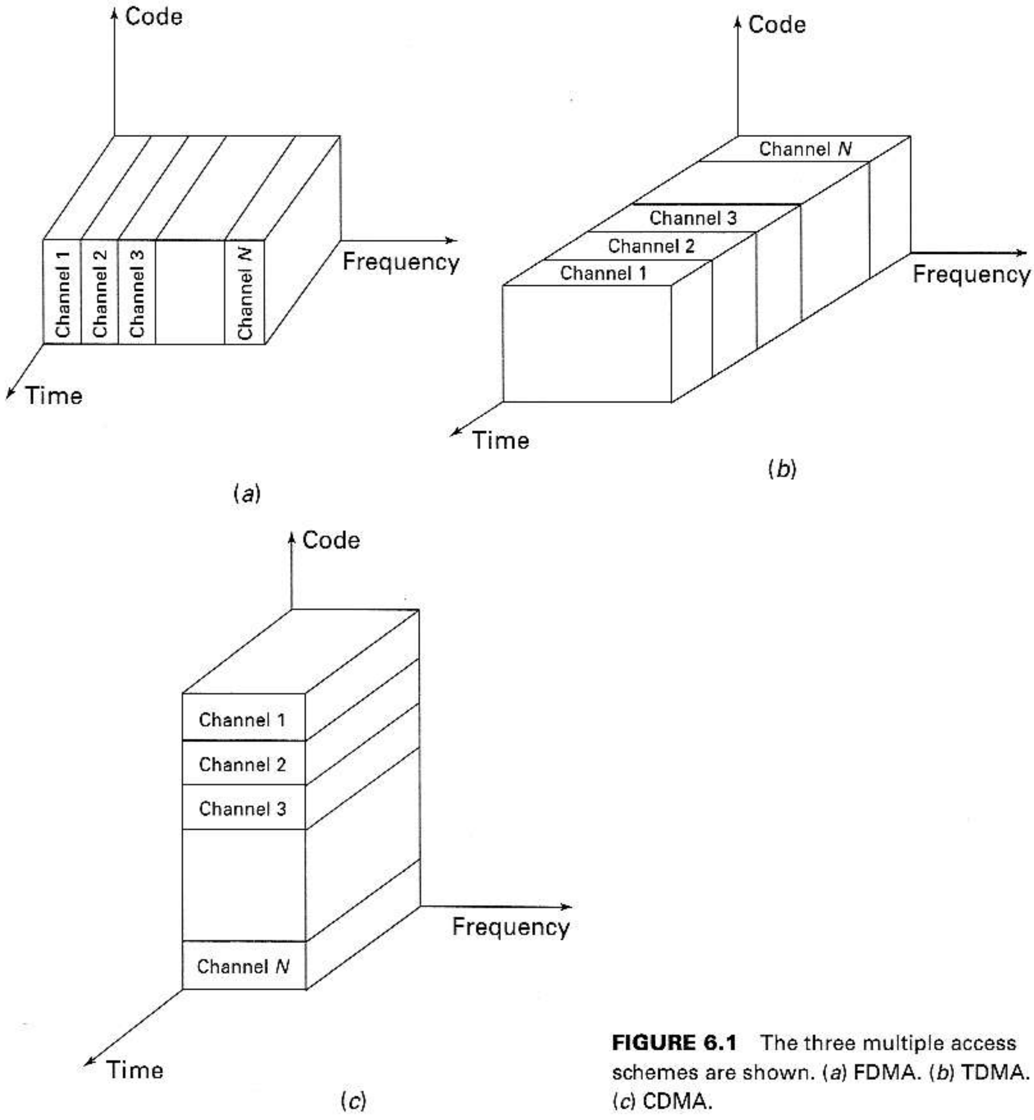


FIGURE 6.1 The three multiple access schemes are shown. (*a*) FDMA. (*b*) TDMA. (*c*) CDMA.

6.1 FREQUENCY DIVISION MULTIPLE ACCESS

FDMA is one of the simplest schemes used to provide multiple access. It easily separates different users by having each user operate at a different carrier frequency. The multiple users can therefore be isolated using bandpass filters. Frequency division multiple access is the mechanism used in analog cellular systems.

The principle of FDMA is shown in Figure 6.4. The available bandwidth W is divided into N nonoverlapping bands, each of width W_{ch}. A small guardband is provided so that interference from adjacent channels will be reduced in the event of any instability in the carrier frequencies of the neighboring channels. When a user makes a call request to the base station, the BS assigns one of the unused channels, which then becomes the exclusive "property" of that particular user, and nobody

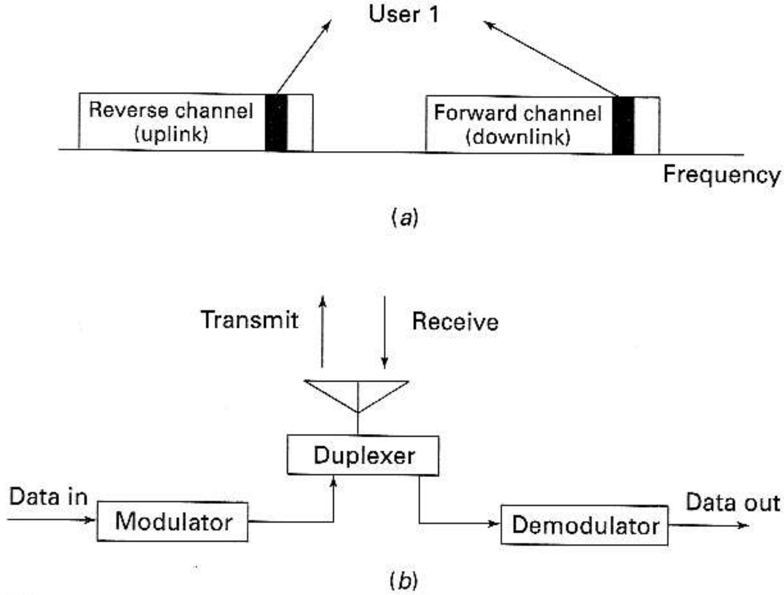


FIGURE 6.2 Concept of frequency domain duplex (FDD).

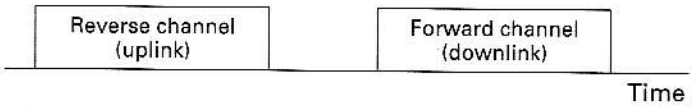


FIGURE 6.3 Concept of time domain duplex (TDD).

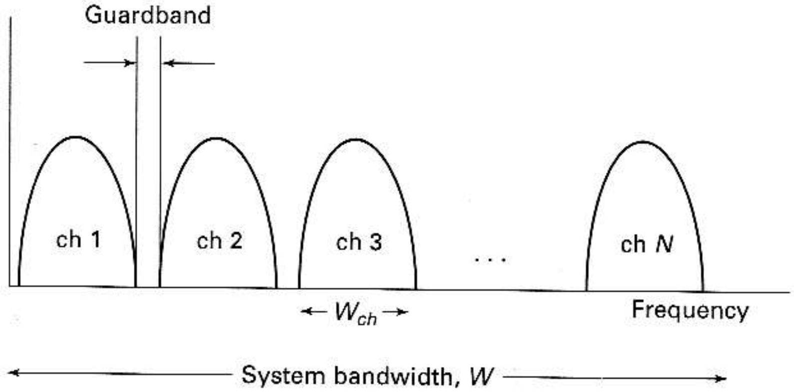


FIGURE 6.4 Principle of frequency division multiple access.

else will be assigned that channel. When the user terminates the call, the frequency may be reassigned to another user. If during the call, the caller moves into another cell, the caller will be assigned an unused channel in the new cell. If frequency domain duplex (FDD) is used, the available band is divided in two; one half is used for the forward channel, and the other half is used for the reverse channel. The caller has one frequency for the forward channel and another frequency for the reverse channel.

The major advantage of the FDMA system is its hardware simplicity, since discrimination between users is managed using simple bandpass filters. No timing information or synchronization is required. Since the bandwidth assigned each user is relatively small, the problems of frequency-selective fading are essentially nonexistent and the fading is purely flat. FDMA systems also have a number of major disadvantages. Let us briefly review them.

If a FDMA channel is not in use, it sits idle and cannot be used to enhance the capacity of the system. This is to say that the idle channel cannot be assigned to another cell unless some form of dynamic channel assignment is possible, in which unused channels may be assigned to the other cells that need more channels. Since the multiple access schemes rely heavily on bandpass filters, these filters must have excellent cutoff characteristics. The major problem in FDMA systems is the cross-talk arising from adjacent channel interference produced by nonlinear effects. The many channels that compose the FDMA system use the same antenna and, therefore, the associated power amplifiers. Since amplifiers have some level of nonlinearity, intermodulation products will result.

Consider a simple example of a three-channel case where the composite signal, $c(t)$, at the receiver can be expressed as

$$c(t) = a_1(t)\cos(2\pi f_1 t) + a_2(t)\cos(2\pi f_2 t) + a_3(t)\cos(2\pi f_3 t), \qquad (6.1)$$

where f_1, f_2, f_3 are the carrier frequencies and $a_1(t)$, $a_2(t)$, $a_3(t)$ are information-bearing signals. The output of a nonlinear amplifier, $c_{\text{out}}(t)$, will be

$$c_{\text{out}}(t) = b_0 + b_1[c(t)] + b_2[c(t)]^2 + b_3[c(t)]^3 + \cdots \qquad (6.2)$$

where the b_i are the scaling factors. Depending on the ratio of the carrier frequencies, the nonlinear terms can result in terms of the type

$$f_1 = 2f_2 - f_3 \qquad (6.3)$$

or any other combination such that signals from other channels will appear in the same frequency window of the signal being received, leading to interchannel interference. An appropriate frequency planning system can reduce the cross-talk induced by intermodulation.

A problem with the FDMA system is its inability to be used in variable-rate transmission, which is becoming common in digital systems. Variable-rate transmission makes it necessary to employ a number of modems at the terminal. This eliminates FDMA as the choice for combined voice and data transmission.

Another drawback of the FDMA system is its inherent need for transmitters and receivers with high Q values to ensure excellent channel selectivity. Monitoring this may also be difficult.

6.2 TIME DIVISION MULTIPLE ACCESS (TDMA)

The TDMA technique enables users to access the whole bandwidth, which is allocated on a time basis (Skla 1988, Naka 1990, Samp 1997). Each user/channel occupies the whole bandwidth for a fraction of the time, called a slot. But if the user continues to have access or reservation to the bandwidth on a periodic or rotational basis, it is possible for the user to carry on the conversation or transmission of information on a nearly continual basis. If there are N users, we have a frame of N time slots, as shown in Figures 6.5 and 6.6.

A conceptual setup of the TDMA scheme is shown in Figure 6.6. The uplink (to the base station) consists of transmission from each user at specific time intervals. The

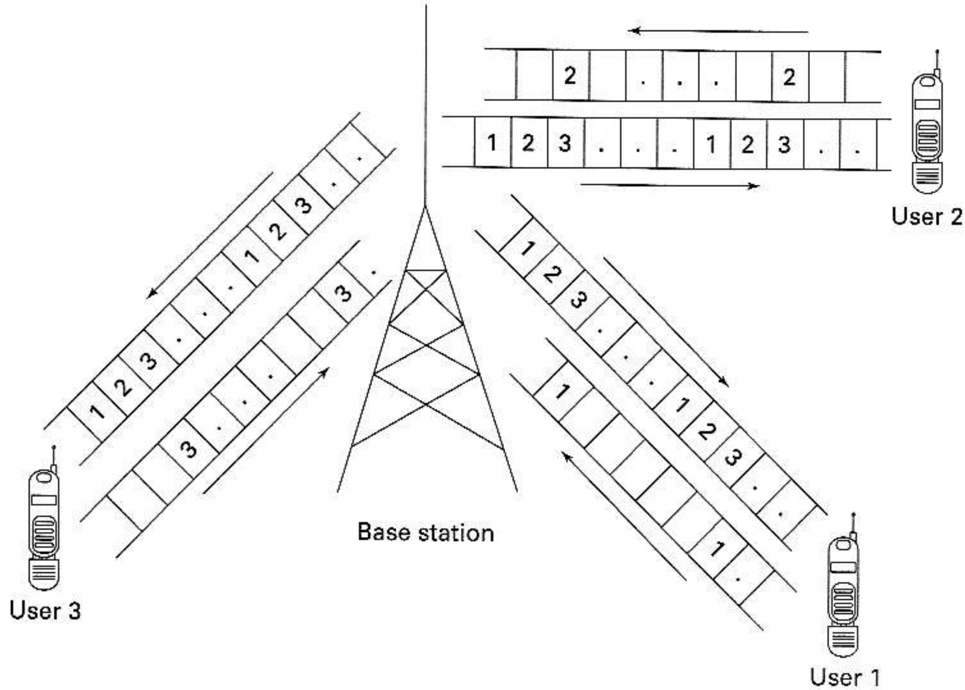


FIGURE 6.5 A number of terminals communicating with a base station in the TDMA scheme.

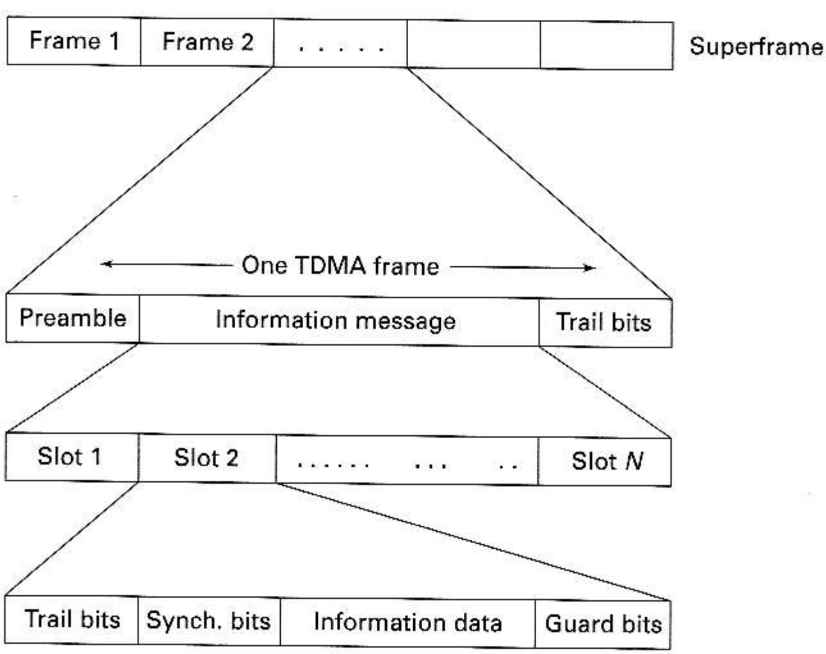


FIGURE 6.6 Details of a complete TDMA frame.

downlink consists of signals corresponding to all the users in the frame. As we can see from Figure 6.6, the uplink consists of signals only at the assigned time slots. A number of frames can be combined to produce a superframe. Each frame is made up of a preamble, an information message, and trail bits. In Figure 6.6, the preamble consists of the address and synchronization information that both the base station and the subscribers can use for identification. If time domain duplex (TDMA/TDD) is used, half the time slots in the frame will correspond to the uplink channels and the other half will correspond to the downlink channels. On the other hand, if frequency domain duplex (TDMA/FDD) is used, the downlink channels and uplink channels will be separate frames, at different carrier frequencies. As shown in Figure 6.5, several time slots of delay are introduced between the uplink and downlink time slots for a particular user.

One of the important issues in TDMA systems is the synchronization. If the time slot synchronization is lost, the channels may collide with each other. To maintain synchronization, the base station periodically sends a frame timing signal, which the MUs use for synchronization. In fact, we need synchronization of the slot, frame, and superframe. While susceptibility of FDMA to fading is not severe, TDMA systems suffer from frequency-selective fading. Processing to overcome the effects of fading must be undertaken. On the positive side, TDMA makes it relatively easy to measure the power levels, and hence to perform hand-off procedures.

6.3 CODE DIVISION MULTIPLE ACCESS

Most of the systems described in the previous chapters achieve multiple access using a frequency division technique, a time division multiple access technique, or a combination of the two. These techniques are used in environments where there is a premium on the available bandwidth. On the other hand, if a large chunk of bandwidth is available, it is possible to utilize this large bandwidth for a single user (Coop 1979; Pick 1982; Shap 1994; Yue 1983; Vite 1979, 1985; Wang 1993). This, however, is an extremely inefficient way of utilizing the available bandwidth. Highly efficient use of the bandwidth can be accomplished if a large number of users can occupy the same bandwidth at all times and if each user is assigned a different set of "bits" or codes. In other words, it is possible to share the bandwidths through code division multiple access or CDMA, where each user is assigned a pseudorandom or pseudonoise (PN) binary-valued sequence or code (MacW 1976).

This distinguishing feature of the CDMA system, namely, the wider bandwidth, involves nothing but spreading of the spectrum of the transmitted signal through the use of much narrower pulses (Scho 1977, 1982; Skla 1988; Taub 1986; Schi 1994; Proa 1995; Wu 1995). Thus the spectrum of the transmitted signal is wider than the spectrum associated with the data rate. To illustrate, the spectrum of the signal associated with the data rate is shown in Figure 6.7 along with the time domain pulse. Now, if we were to transmit a set (K) of extremely narrow pulses during the period T, with randomly chosen values of amplitude (+ or -1), each of duration $T_c = T/K$, the spectrum of the transmitted signal would become wider. The duration of these narrow pulses, or chips, as they are commonly referred to, is T_c. This spreading of the spectrum, or the spread-spectrum approach, is the key to implementation of the CDMA technique.

The basic principle of the CDMA technique is illustrated in Figure 6.8. The input bit stream and the chip sequence are shown. The number of chips/bit is K. Once the data sequence is multiplied by the chip sequence, the spectrum of the information is spread.

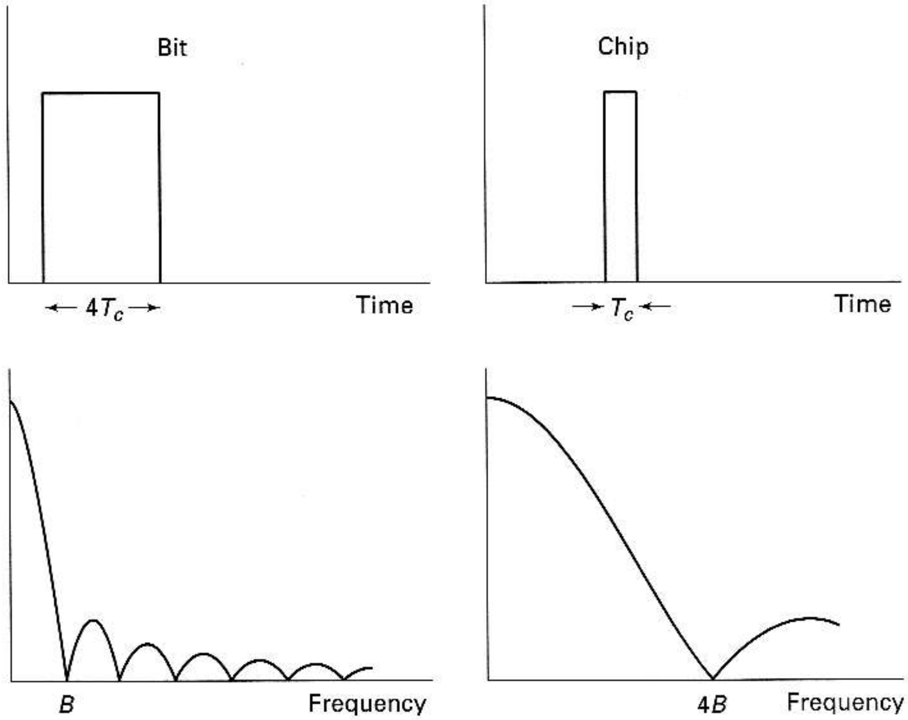


FIGURE 6.7 A bit of duration $T = 4T_c$ and a chip of duration T_c are shown along with their spectra. In this figure, $K = 4$.

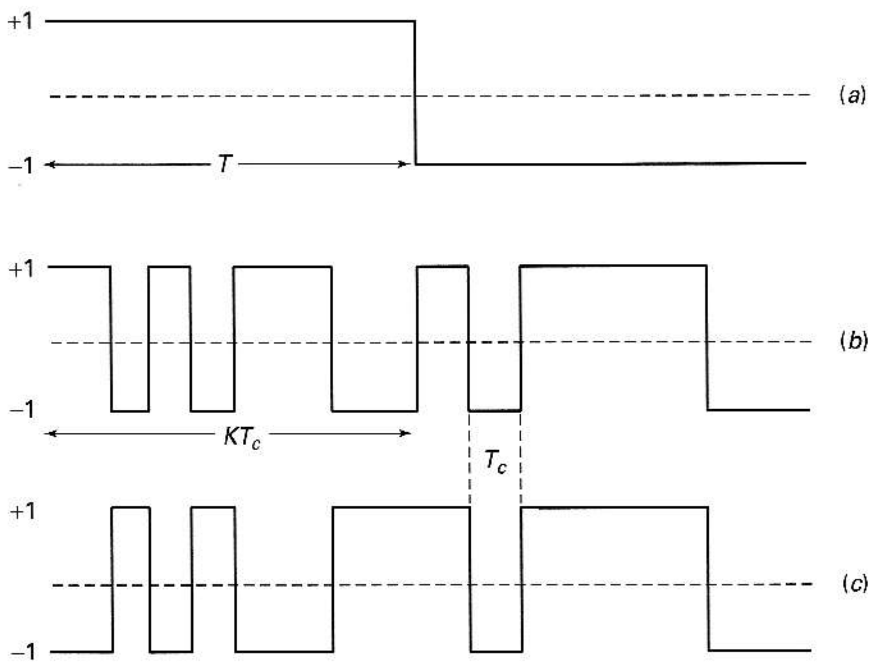


FIGURE 6.8 (*a*) Input data stream. (*b*) Chip sequence. (*c*) Modulo-2 addition of the bit stream and the chip sequence.

The PN sequence is unique to each user and is almost orthogonal to the sequences of other users. Thus, the interference from other users in the same band will be much less. The number of orthogonal codes, however, is limited. As the number of users increases, the codes become less and less orthogonal and the interference increases. This allows a soft capacity limit for allowing new users into the system. As more and more users enter the system, the quality of the channel goes down, but nobody needs to be turned away. This also makes the frequency planning and design of cells relatively easy, since all users are using the same frequency band. This is in sharp contrast to systems based on TDMA/FDMA, where there is a hard capacity limit and a need for strict frequency planning and cell design.

In addition to the advantages of soft capacity and the relative ease of frequency planning, the use of very short pulses provides an easy means to combat the effects of multipath fading. In non-CDMA systems, the multipath effects lead to broadening of the pulses owing to the nonresolvable paths. In CDMA, the pulses are very narrow, and therefore the multipath fading produces nonoverlapping, resolvable pulses at the receiver, corresponding to the distinct paths. This resolvable multipath scenario is akin to multipath diversity since each of these resolvable paths corresponds to a different branch of the diversity system. These nonoverlapping pulses can be combined to combat the effects of fading using a RAKE receiver, which is nothing more than a means of combining the multipath pulses with appropriate delays and weights, as we will see later in this chapter.

The spread-spectrum system may use either of the two modulation schemes PSK or FSK. The PN sequence in conjunction with PSK generates phases of 0 and π pseudorandomly in accordance with the code, at a rate that is an integral multiple of the data rate. This approach results in direct-sequence systems (DS-CDMA). On the other hand, if the PN sequence is used in conjunction with M-ary FSK to select the frequency of the transmitted signal pseudorandomly (Skla 1988, Pick 1982), the result is a frequency-hopped direct-sequence system (FH-CDMA).

6.3.1 Description of a PN Code Generator

The pseudonoise (PN) sequence is a periodic binary sequence that appears noiselike. It can be generated using a feedback shift register, as shown in Figure 6.9. The setup consists of a number of flip-flops (two-state memory stages) and a logic element, typically a modulo-2 adder, interconnected to form a feedback circuit. A single timing clock regulates all of the flip-flops.

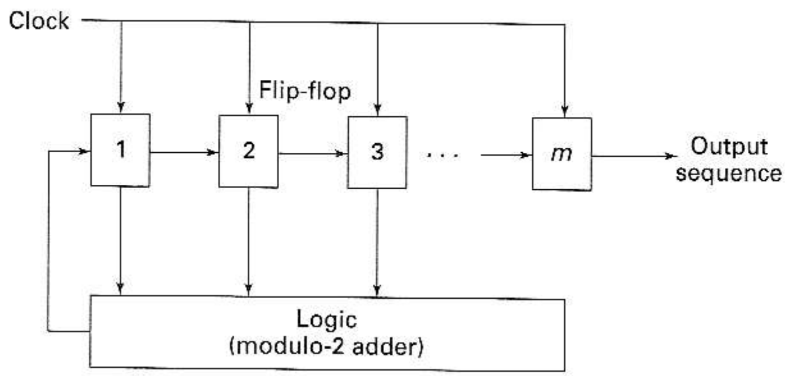


FIGURE 6.9 Block diagram of a PN code generator.

6.3.2 Properties of Pseudonoise or Pseudorandom Sequences

A PN sequence is a collection of positive and negative 1s, in almost equal numbers, occurring randomly. Using the feedback shift register of m stages shown in Figure 6.9, we expect to have $2^m - 1$ sequences of numbers, also known as a maximal-length sequence. The sequence repeats itself, and thus has a period of $2^m - 1$.
Some of the properties of the maximal-length sequences are as follows.

- In each period of a maximal-length sequence, the number of +1s is exactly one more than the number of −1s. This is known as the *balance property.*
- The autocorrelation function of a maximal-length sequence is binary valued and periodic.

Even though the sequence is deterministic and, therefore, not completely random, it appears like white noise to an unauthorized/unknown user. The autocorrelation of the code can therefore be identified with the autocorrelation of band-limited white noise. A typical code sequence and its autocorrelation are shown in Figure 6.10. Figure 6.10*a* shows the autocorrelation, $R(\tau)$, of white noise, and Figure 6.10*b* shows the autocorrelation, $R_{\text{PN}}(\tau)$, of the PN code. We can see that the code decorrelates with shifted versions of the code, as evidenced by the fast dropoff of the autocorrelation. Note that when the code length becomes infinite, the autocorrelation of the PN code and that of the band-limited white noise become equal.

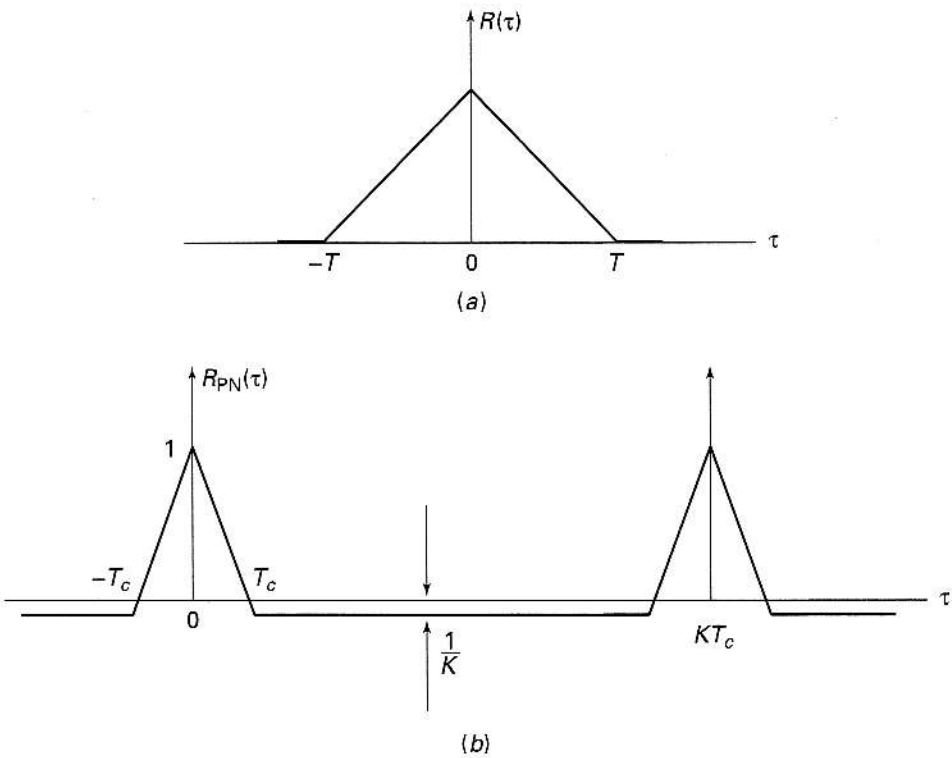


FIGURE 6.10 (*a*) Autocorrelation of a random sequence. (*b*) Autocorrelation of a PN sequence of length K. The chip duration is T_c.

This decorrelation property of the code makes it possible to use time diversity to combine delayed versions of the signals when a multipath exists. We will examine this aspect in Section 6.3.4.

6.3.3 Direct-Sequence Spread-Spectrum Modulation

The direct-sequence spread-spectrum modulated signal (Skla 1988; Taub 1986; Pick 1982, 1991; Pado 1994) can be generated using the setup shown in the block diagram of Figure 6.11. The bit duration is T and the chip duration is T_c, where a "chip" is identified as a single pulse of the PN waveform. The ratio of T to T_c gives the number of chips in a bit. The encoded data are added to the PN code chips in a modulo-2 fashion before being modulated using a BPSK scheme.

The transmitted signal, $s_{\text{DSSS}}(t)$, can be expressed as

$$s_{\text{DSSS}}(t) = \sqrt{\frac{2E}{T}} m(t)p(t)\cos(2\pi f_0 t + \theta)\,, \tag{6.4}$$

where $m(t)$ is the encoded data, $p(t)$ is the PN chip sequence, and θ is the phase at $t =$ 0. Note that $m(t)$ consists of data symbols (± 1) of duration T while $p(t)$ consists of chips (± 1) with a chip duration of T_c, with $T_c \ll T$, with the provision that $T = KT_c$, where K is an integer. The very short duration of the chips also means that the bandwidth of the DSSS signal is K times the bandwidth of conventional BPSK, where the symbol duration is T. This increase in bandwidth is shown in Figure 6.7. We will come back to the issue of increased bandwidth when we discuss interference suppression in DS-CDMA systems.

The receiver for the DSSS system (CDMA-R) is shown in Figure 6.12. The received signal is multiplied by the PN code and filtered. This, in effect, results in a correlator. The output is passed through a BPSK demodulator to recover the original data.

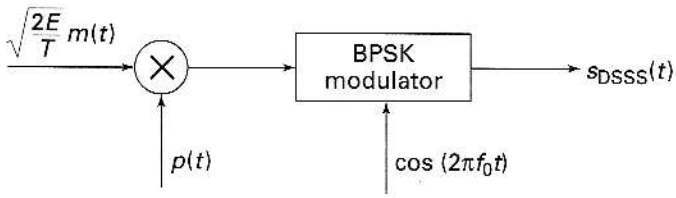


FIGURE 6.11 DS-CDMA transmitter block diagram.

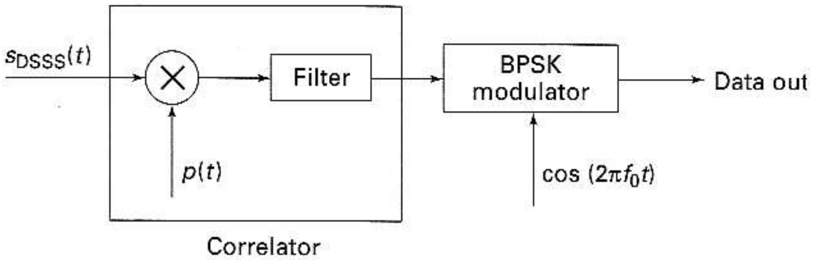


FIGURE 6.12 Block diagram of a DS-CDMA receiver.

The question arises about the expected probability of error in this form of modulation/demodulation. The multiplication by the PN code $p(t)$ at the transmitter, and later at the receiver stage, results in no change in the signal level since the product is always unity. Thus, the overall performance of the DSSS system in an ideal case should have no effect from the presence of the PN code as far as thermal noise is concerned. The error probability is once again given by the error probability for the BPSK receiver, eq. (3.83):

$$p_{\mathrm{DSSS}}(e) = p_{\mathrm{BPSK}}(e) = \frac{1}{2}\mathrm{erfc}\left(\sqrt{\frac{E}{N_0}}\right). \tag{6.5}$$

Even though the performance of the DSSS system is no better than that of a pure BPSK system in the presence of thermal noise, the DSSS system has an exceptional ability to suppress in-band interference. This can be understood by treating the interference as akin to noise. Consider the case of a single-tone interference. The input to the receiver, $c_{\mathrm{in}}(t)$, can be written as

$$c_{\mathrm{in}}(t) = m(t)p(t)s(t) + A_{\mathrm{int}}s(t)\,, \tag{6.6}$$

where the second term is the interfering term, with A_{int} being a scaling factor. In eq. (6.6),

$s(t)$ is a BPSK carrier signal.

$m(t)$ is the bipolar data.

$p(t)$ is the PN code.

For A_{int} equal to 1, the powers of the signal of interest and the interferer are the same. If A_{int} is larger than 1, the interfering signal will be stronger, and this will lead to problems. The issue of this "near/far problem" will be discussed later in this chapter. The spectra of the signal and interferer are shown in Figure 6.13.

Note that the interfering term has a much smaller bandwidth since it has not been multiplied by the PN code, $p(t)$, while the signal spectrum is broad. Input $c_{\mathrm{in}}(t)$ is first multiplied by the code $p(t)$ and then applied to a BPSK demodulator. The input to the modulator can be expressed as

$$c_{\mathrm{in}}(t) = \text{signal} + \text{noise}\,, \tag{6.7}$$

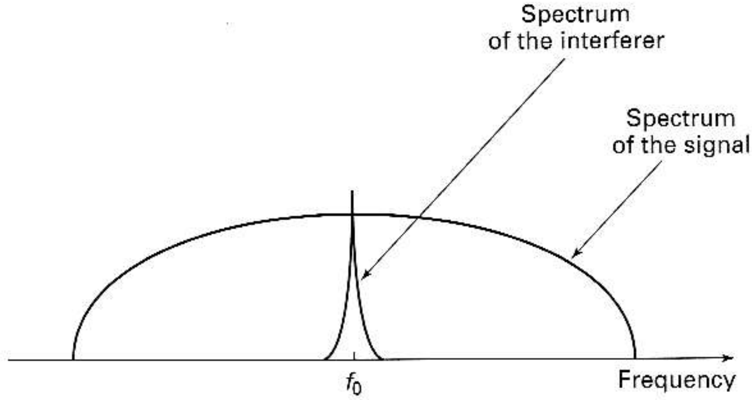


FIGURE 6.13 Spectrum of the signal along with the spectrum of the interferer during transmission.

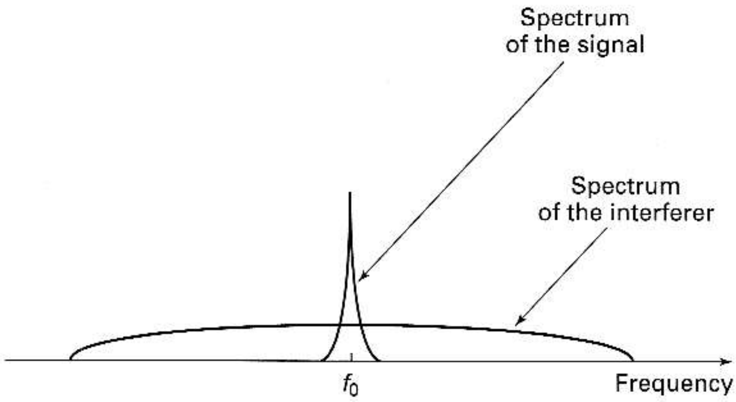


FIGURE 6.14 Spectrum of the signal along with the spectrum of the interferer after correlation.

where the signal term is identical to the signal component at the output of the BPSK receiver (since $p(t) \times p(t) = 1$) and the noise is given by

$$\text{Noise} = A_{\text{int}} p(t) n_{\text{int}}(t) . \tag{6.8}$$

The value $n_{\text{int}}(t)$ is the interfering signal that has the same power as the primary signal. Note that the spectrum of the *signal* now is narrower, while the spectrum of the *noise* is broader because of the presence of the spreading code $p(t)$. The bandwidth of the *noise* is now K times the bandwidth of the *signal* component. This is shown in Figure 6.14.

We can now calculate the error probability by treating the second term in eq. (6.6) as noise (neglecting thermal noise):

$$p(e) = \frac{1}{2}\operatorname{erfc}\sqrt{\frac{E}{N_0}} = \frac{1}{2}\operatorname{erfc}\sqrt{\frac{P_s T}{P_{\text{int}} T_c}} = \frac{1}{2}\operatorname{erfc}\sqrt{\frac{K}{A_{\text{int}}^2}}, \tag{6.9}$$

where P_s is the signal power and P_{int} is the interfering signal power. As the value of K increases, the length of the code increases and the error from interference decreases. We can therefore regard K as the processing gain from the use of the spread-spectrum technique. For $A_{\text{int}} = 1$ (equal power of the interfering signal), the error probability becomes

$$p(e) = \frac{1}{2}\operatorname{erfc}\sqrt{K} . \tag{6.10}$$

We can now consider the case of DS-CDMA. The block diagram of the receiver is shown in Figure 6.15. The length of the code is once again KT_c, and we consider the case where there are k users in a given frequency band. We also assume that the k codes are almost uncorrelated with one another. The composite received signal, $c_k(t)$, at any one of the receivers will be

$$c_k(t) = \sum_{i=1}^{k} m_i(t) p_i(t) s(t) , \tag{6.11}$$

where it has been assumed that all the k signals, identified by $m_i(t)$, and the corresponding PN codes, $p_i(t)$, have the same energy. The carrier signal, $s(t)$, is $\cos(2\pi f_0 t)$.

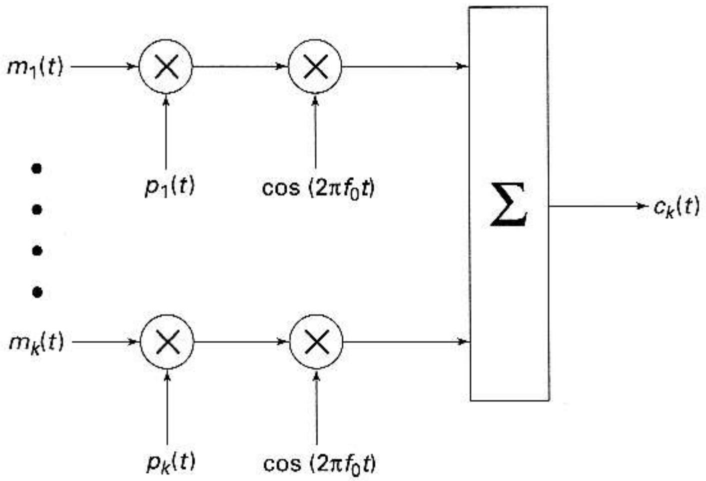


FIGURE 6.15 DS-CDMA receiver showing all the multiple channels.

Consider the case of a receiver of the first channel ($k = 1$). The output, $c_{k1}(t)$, of the multiplier with $p_1(t)$ will be

$$c_{k1}(t) = \sum_{i=1}^{k} m_i(t)p_i(t)p_1(t)s(t). \tag{6.12}$$

Once again, this signal will be the input to a BPSK demodulator, as shown in Figure 6.16. The input, $c_{k1\text{in}}(t)$, to the demodulator can now be expressed as

$$c_{k1\text{in}}(t) = \text{signal} + \text{noise} \tag{6.13}$$

where the signal comes from the first term of the summation in eq. (6.12) and the noise represents all the other ($k - 1$) terms, given by

$$\text{Noise} = \sum_{i=2}^{k} m_i(t)p_1(t)p_i(t). \tag{6.14}$$

Assuming that all the codes are nearly uncorrelated, we can see that there will be $k - 1$ interfering components, each with the same power as the signal, and, hence, the error probability from the interfering signals becomes

$$p(e) = \frac{1}{2}\text{erfc}\left(\sqrt{\frac{K}{k-1}}\right). \tag{6.15}$$

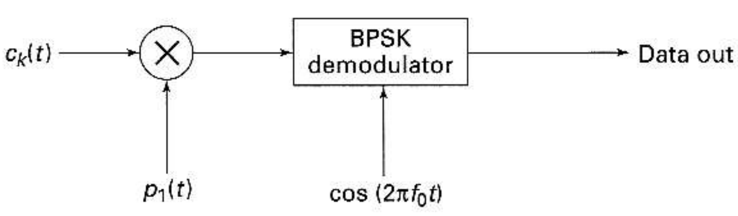


FIGURE 6.16 Demodulator.

Note, however, that it is possible to have the powers of the interfering signals be higher than the power of the signal of interest being received, creating significant variation in the performance.

EXAMPLE 6.1

In a DS-CDMA cell, there are 24 equal power channels that share a common frequency band. The signal is being transmitted using a BPSK format. The data rate is 9600 bps. A coherent receiver is used for recovering the data. Assuming the receiver noise to be negligible, calculate the chip rate needed to maintain a bit error rate of 10^{-3}.

Answer Assuming there is no thermal noise, the bit error rate is given by eq. (6.15), where K is the processing gain and k is the number of channels. BER $= 10^{-3} = 0.5\ \text{erfc}(\sqrt{z})$, where $z = K/(K-1)$. Using the MATLAB function *erfinv(.)*, we can solve for $z = [erfinv(1-2^{*}\text{BER})]^2$.

We get $z = 4.77$. We are given $k = 24$; $K = 23 \times 4.77 = 109.82$. Since $K =$ (chip rate)/(data rate), chip rate $= 109.82 \times 9600 = 1.05$ Mchips/s. ■

EXAMPLE 6.2

A DS-CDMA system is expected to have a processing gain of 30 dB. The expected data rate is 9600 bps. What should the chip rate be? If BPSK modulation will be employed, what is the bandwidth required for transmission using a null-to-null criterion?

Answer Processing gain $= 30$ dB $= 10\log_{10}(K)$; $K = 1000$; data rate $= 9600$ bps

$$\text{Chip rate} = K \times \text{data rate} = 9{,}600{,}000 = 9.6 \text{ Mchips/s.}$$

$$\text{BW} = 2 \times 9.6 \text{ MHz} = 19.2 \text{ MHz.}$$

■

6.3.4 RAKE Receiver in DS-CDMA Systems

The multipath effects present in wireless systems adversely impact the TDMA-based systems since the multiple pulses arriving at the receiver are nonresolvable (see Chapter 2). These pulses thus overlap, leading to broadening of the pulse, resulting in frequency-selective fading as discussed in Chapter 2. However, the chip duration in DS-CDMA systems is very narrow, and the delays between multiple paths may be larger than the chip duration. Under this condition, the delayed versions of the chips are resolvable, providing time diversity (Skla 1993; Chan 1994a; Pick 1982, 1991; Kohn 1995; Proa 1995; Pani 1996). The fact that the PN sequences have very low correlation makes it possible to separate these delayed versions and separately perform the correlation. The block diagram in Figure 6.17 shows the concept of a RAKE receiver implementation for DS-CDMA systems.

The different correlators (see Chapter 3 for a discussion of correlators) can be synchronized to various paths with different delays and programmed to capture the strongest signals coming from multipath components. Note that the signal arriving at any given time is synchronized with one of the correlators and, as a result, will have negligible correlation with the other two correlators by virtue of the low correlation of the code. The outputs from the correlators are appropriately weighted and combined for decisionmaking and recreation of the data bits.

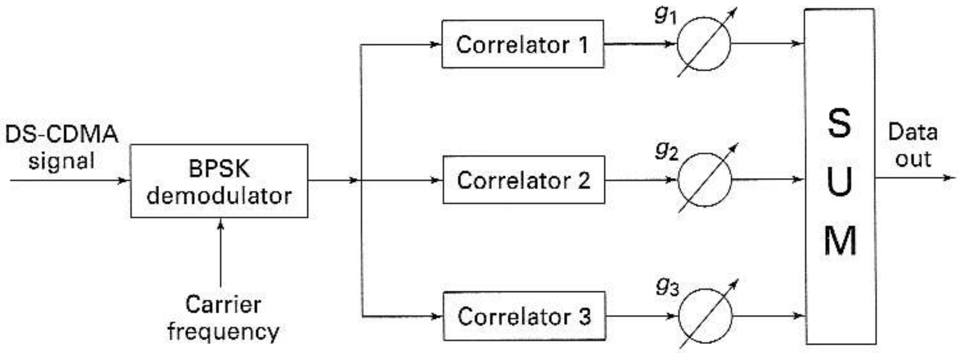


FIGURE 6.17 Conceptual block diagram of a RAKE receiver. Three multiple paths are arriving and being processed in three separate correlators. Each output is weighted by a factor g_i, $i = 1, 2, 3$.

EXAMPLE 6.3

A RAKE receiver is being designed to take advantage of the multipath effects in the channel. If the minimum delay difference is 300 m, what is the minimum chip rate necessary to successfully resolve the multipath components and operate the RAKE receiver?

Answer If the chip rate is not sufficient, the multiple paths corresponding to the chips will not be resolvable, and the requirement of separable pulses will not be met. This means that the chip duration must be smaller than

$$\tau = \frac{\text{delay distance}}{\text{speed of the e.m. wave}} = \frac{300}{3 \times 10^8} = 1\ \mu\text{s}$$

The chip rate must be greater than $1/\tau = 1$ Mchip/s. ■

6.3.5 Frequency-Hopping Spread-Spectrum Technique

While the direct-sequence spread-spectrum technique uses phase modulation (and may also be viewed as amplitude modulation), the frequency-hopping spread-spectrum technique uses frequency modulation. In fact, frequency hopping involves the hopping of the carrier frequency in a random fashion. The set of possible frequencies used is referred to as a *hopset* (Skla 1988, Taub 1986, Kohn 1995). The starting point of a FH-SS technique is either a BFSK signal or a MFSK signal. If one considers a conventional BFSK signal, the carrier frequency is alternated between two *fixed* frequencies. In a FH/BFSK system, the data symbol modulates a carrier whose frequencies are *pseudorandomly* determined. This statement can be understood with the aid of the block diagram of a FH/BFSK modulator shown in Figure 6.18.

The output of a conventional BFSK modulator and the output from a digital frequency synthesizer are applied to a mixer. The bandpass filter selects the sum frequency coming out of the product modulator. The successive k-bit segments of the PN sequence drive the frequency synthesizer, enabling the carrier frequency to hop over 2^k distinct values. For a given hop, the bandwidth of the FH/BFSK signal is the same as for conventional BFSK. Over many hops (2^k), the bandwidth occupied by the transmitted signal will be orders of magnitude higher than for conventional BFSK. Indeed, it is possible to have a spectral spreading in FH/BFSK far exceeding that

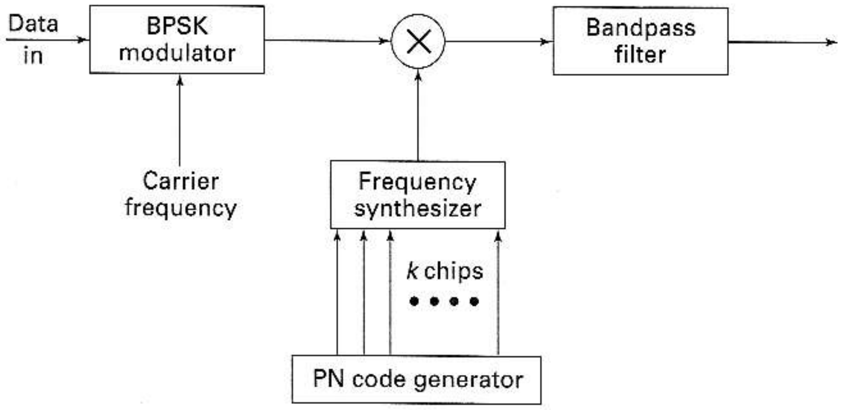


FIGURE 6.18 FH-CDMA transmitter.

observed in direct-sequence techniques. Consequently, the processing gain in FH/BFSK systems can be superior to that in DS-SS systems.

The rate at which the frequency hops determines whether the system is a slow-hopping or a fast-hopping system. We have *slow frequency hopping* (SFH) if the symbol rate, R_s, is higher than the hopping rate, R_h. This means that several symbols will be transmitted on each frequency hop. If, on the other hand, R_h is higher than R_s, we have *fast frequency hopping* (FFH), where the carrier frequency will change several times during the transmission of a single symbol. The term *chip* is defined differently in the frequency hopping context. In the DS systems, a "chip" refers to the shortest duration. In FH systems, a "chip" refers to the shortest uninterrupted waveform in the system. The chip rate, R_c, for a FH system is

$$R_c = \max[R_h, R_s]. \tag{6.16}$$

Figure 6.19 demonstrates the difference between FFH and SFH. In Figure 6.19*a*, we see that the symbol rate is 20 symbols/s while the frequency hopping rate is 40 hops/s. The chip rate here is the hop rate, since it is the maximum of the two (1 chip = 1 hop). This is FFH. In Figure 6.19*b*, the data rate is still 20 symbols/s but the hopping rate is reduced to 6.66 hops/s. In this case, changes in the waveform are due to modulation, and therefore, the chip rate is the symbol rate (1 chip = 1 symbol) since the data symbol is shorter than the hop duration. This is an example of SFH.

A typical demodulator setup for frequency-hopping spread-spectrum is shown in Figure 6.20. Since the demodulator is essentially a noncoherent BFSK receiver, the performance of FH spread-spectrum systems can be evaluated. The probability of error, $p(e)$, associated with a noncoherent BFSK receiver is given by

$$p(e) = \frac{1}{2}\exp\left(-\frac{E}{2N_0}\right). \tag{6.17}$$

6.3.6 Comparison of DS and FH Systems

The bandwidth of DS systems is related to the PN sequence clock rate or chip rate. The bandwidth of FH systems depends on the tuning range of frequencies and can, with ease, be hopped over a wide bandwidth. The synchronization/timing becomes very crucial in DS systems because of the extremely small chip duration. The timing is less critical in FH systems since hop rates range up to several thousands/s compared with Mbit/s rates

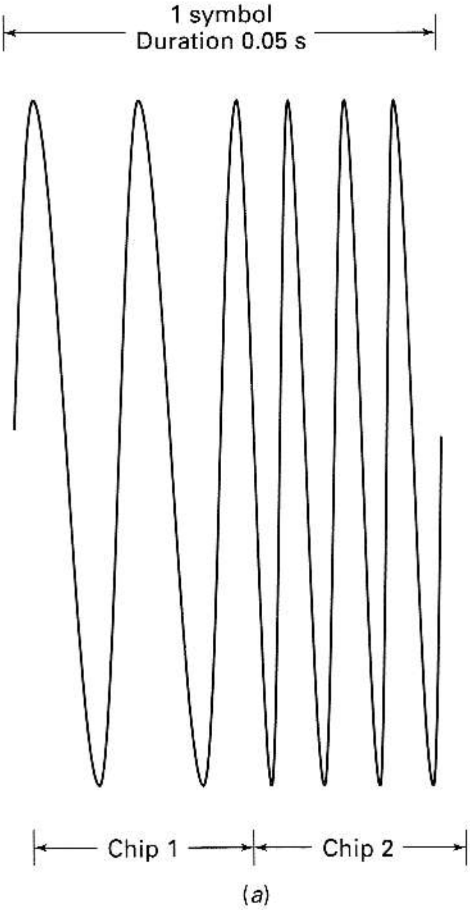
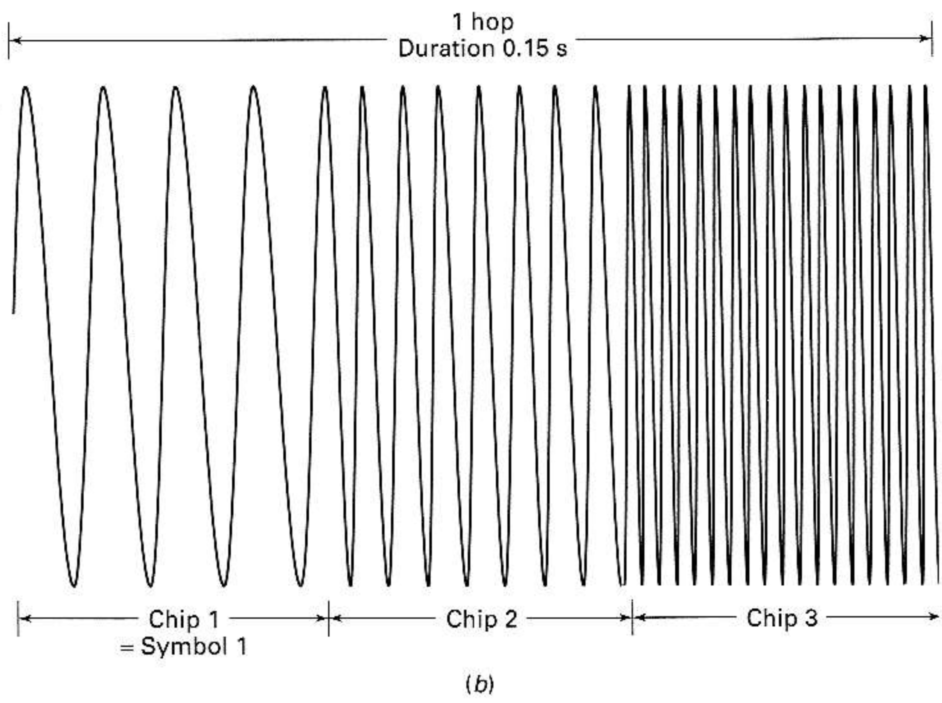


FIGURE 6.19 (*a*) Fast frequency hopping. There are two hops (chip 1 and chip 2) for each symbol. (*b*) Slow frequency hopping. There are three symbols in a hop.



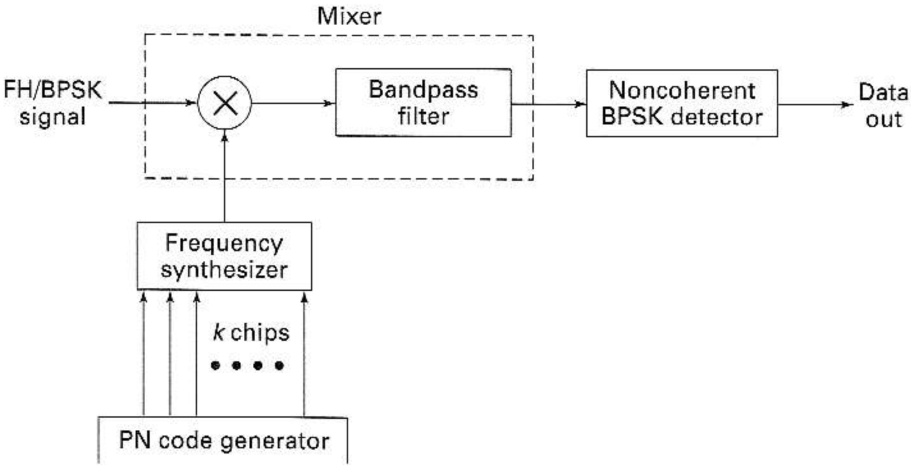


FIGURE 6.20 FH-CDMA demodulator.

in DS systems. The spectrum of the DS system always appears very wide, as it should, while the spectrum of the FH systems is narrow; but the FH spectrum center frequency is changed several times. DS and FH systems are ideal candidates for multipath diversity. The DS system uses extremely short chips, resulting in time diversity. The FH system has an inherent frequency diversity present in fast hopping, with many frequencies used over one data period. While near/far problems are more likely to occur in DS systems because of sharing of the same frequency/bandwidth, they are less likely to occur in FH systems because of the different frequencies.

6.4 OVERVIEW OF WIRELESS SYSTEMS AND STANDARDS

We will now briefly compare the characteristics and features of the major wireless communication systems: AMPS, North American Digital Systems, IS 54 and IS 95, Japanese Digital Cellular Systems, and the Pan-European GSM system.

6.4.1 Advanced Mobile Phone Systems (AMPS)

The first-generation mobile communication systems based on AMPS use frequency modulation for the transmission of signals. For the reverse link (uplink) from the MU to the base station, the frequency band of 824–849 MHz is used. For the forward link (downlink) from the base station to the MU, the frequency band of 869–894 MHz is used. Between the uplink and downlink communication in simplex mode, a separation of 45 MHz exists between radio channels. This large separation permits the use of low-cost duplexers. The maximum frequency deviation of the FM modulator is ±12 kHz. The control channel transmissions and blank-and-burst data streams are transmitted in FSK mode at 10 kbps with a maximum frequency deviation of ±8 kHz. Each base station transmits control data in FSK mode on the forward control channel (FCC) at all times, allowing the MU to lock onto the strongest FCC wherever it is. The base station reverse control