

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 supports multiple users by having each user operate at a different carrier frequency. The rates different 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/N . 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

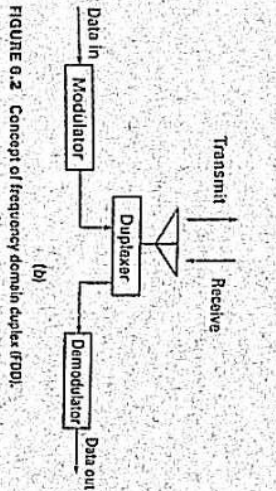
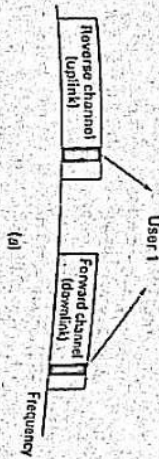


FIGURE 6.2 Concept of frequency domain duplex (FDD).

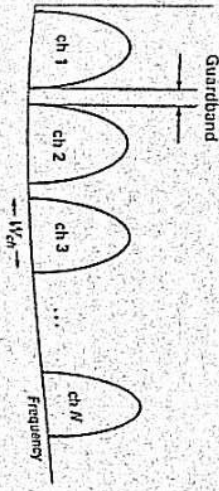


FIGURE 6.4 Principle of frequency division multiple access. When the user terminates the call, the frequency slice will be assigned that channel. If during the call, the caller moves into another cell, the caller will be assigned an unused channel in the new cell. If frequency domain duplex (FDD) is 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 due to 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 user 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

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

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

$$c_{\text{out}}(t) = b_0 + b_1|c(t)|^2 + b_2|c(t)|^3 + \dots \quad (6.2)$$

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

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

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 (Sklar 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 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

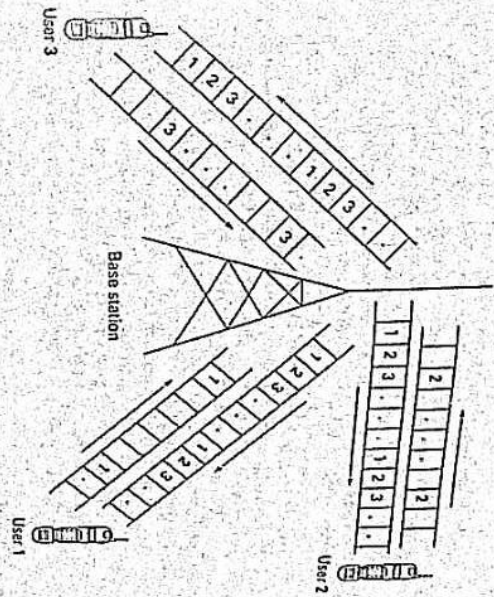


FIGURE 6.5 A number of terminals communicating with a base station in the TDMA scheme.

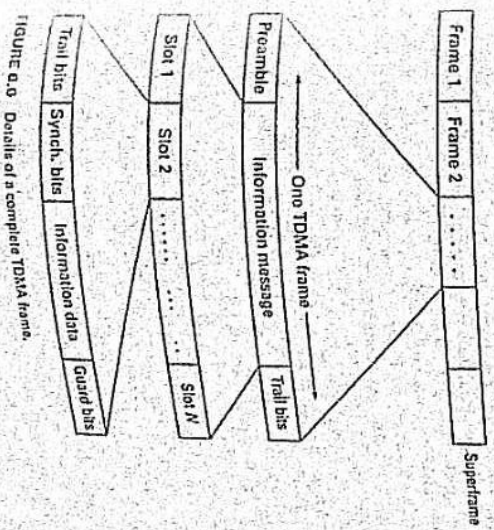


FIGURE 6.6 Details of a complete TDMA frame.

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. As we can see from Figure 6.6, the uplink consists of signals only at the assigned time slots. As we can see from Figure 6.6, the uplink consists of signals only at the assigned time slots. As we can see from Figure 6.6, the uplink consists of signals only at the assigned time slots. As we can see from Figure 6.6, the uplink consists of signals only at the assigned time slots. As we can see from Figure 6.6, the uplink consists of signals only at the assigned time slots.

6.3 CODE DIVISION MULTIPLE ACCESS

Most of the systems described in the previous chapters achieve multiple access using the 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 1983; Ship 1994; Yee 1983; Vile 1979; 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 bandwidth, through code division multiple access or CDMA, where each user is assigned a pseudorandom or pseudonoise (PN) binary-valued sequence or code (Nixey 1976). This distinguishing feature of the CDMA system, namely, the wide bandwidths involves nothing but spreading of the spectrum of the transmitted signal through the use of such narrow pulses (Scho 1977, 1982; Sha 1988; Taub 1986; Shi 1991; Pan 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 (N) of extremely narrow pulses during the period T with randomly chosen values of amplitude (+1 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 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 per bit is K . Once the bit stream is multiplied by the chip sequence, the spectrum of the information is spread

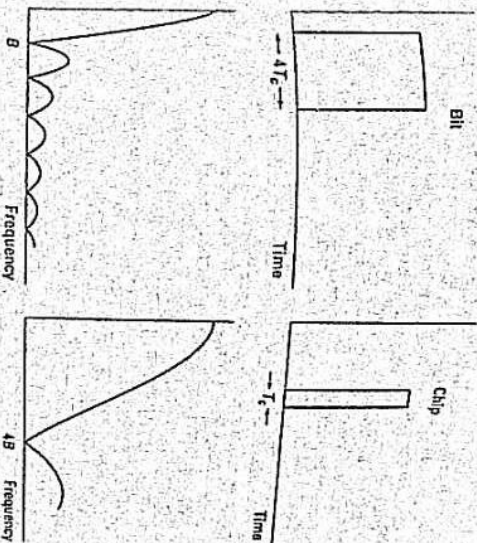


FIGURE 6.7 A bit of duration $T = 4T_c$ and a chip of duration T_c are shown along with their spectra. In this figure, $K = 4$.

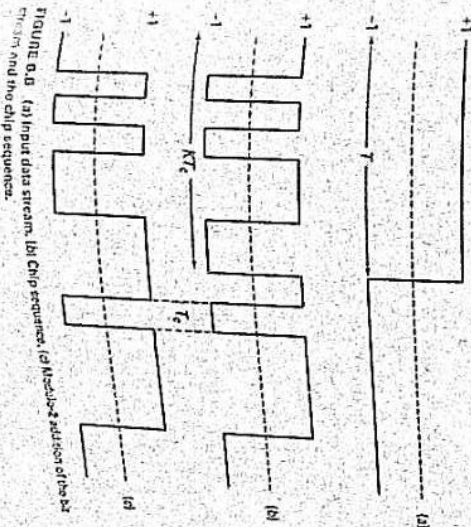


FIGURE 6.8 (a) Input data stream, (b) Chip sequence, (c) Resulting chip sequence after the bit stream and the chip sequence.

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 small. 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/RTMA, 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-SSMA). 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 (SSMA 1988, Peck 1982), the result is a frequency-hopped direct-sequence system (FH-SSMA).

6.3.1 Description of a PN Code Generator

The pseudonoise (PN) sequence is a periodic binary sequence that appears noise-like. 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.

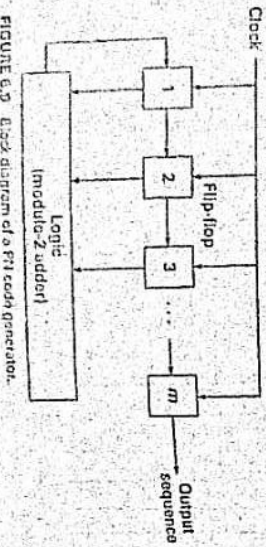


FIGURE 6.9 Clock diagram of a PN code generator.

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 6.10a shows the autocorrelation, $R_{PN}(\tau)$, of white noise, and Figure 6.10b shows the autocorrelation, $R_{PN}(\tau)$, 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.

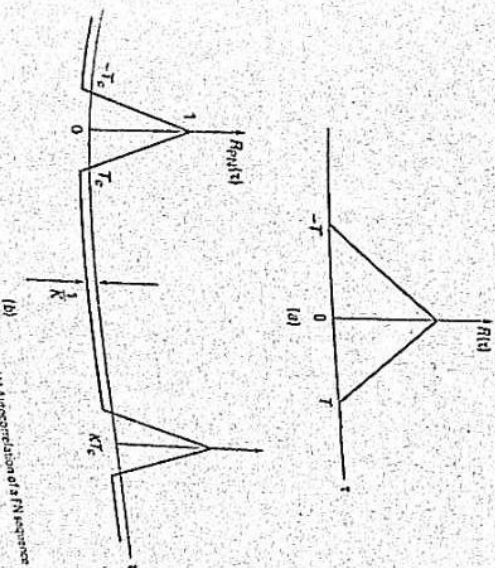


FIGURE 6.10 (a) Autocorrelation of a random sequence. (b) Autocorrelation of a PN sequence of length K . The chip duration is T_c .

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 1996; Pro 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 (XOR) before being modulated using a BPSK scheme.

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

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

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 $T = K T_c$, 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-SS 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.

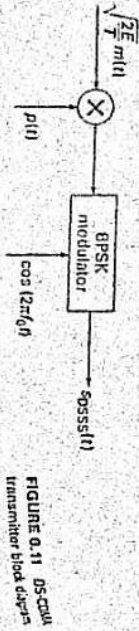


FIGURE 6.11 DSSS transmitter block diagram

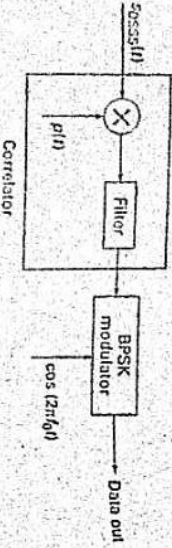


FIGURE 6.12 Block diagram of a DSSS receiver

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 a BPSK receiver, eq. (3.82):

$$P_{\text{DSSS}}(e) = P_{\text{BPSK}}(e) = \frac{1}{2} \operatorname{erfc} \left(\sqrt{\frac{E}{N_0}} \right) \quad (6.5)$$

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, $c_{\text{in}}(t)$, can be written as

$$c_{\text{in}}(t) = m(t)p(t)z(t) + A_{\text{int}}z(t), \quad (6.6)$$

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

$z(t)$ 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 $c_{\text{in}}(t)$ is first multiplied by the code $p(t)$ and then applied to a BPSK demodulator. The input to the demodulator can be expressed as

$$c_{\text{in}}(t) = \text{signal} + \text{noise}. \quad (6.7)$$

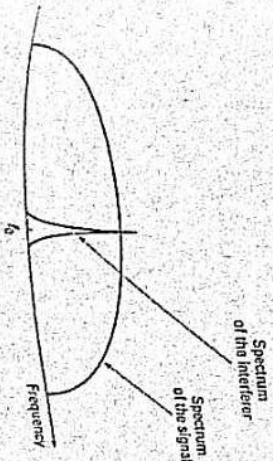


FIGURE 6.13 Spectrum of the signal along with the spectrum of the interferer during transmission.

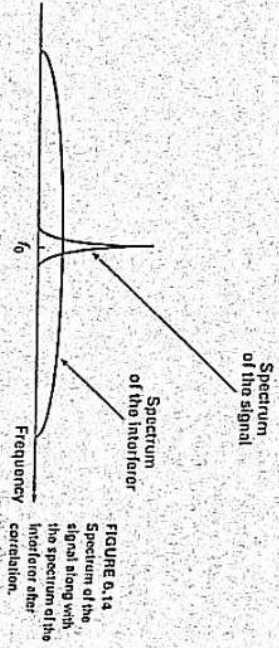


FIGURE 6.14 Spectrum of the signal along with the spectrum of the interferor after correlation.

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

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

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 $p_k(t)$. 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):

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

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

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

We can now consider the case of DS-SSMA. The block diagram of the receiver is shown in Figure 6.15. The length of the code is once again KT , 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

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

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

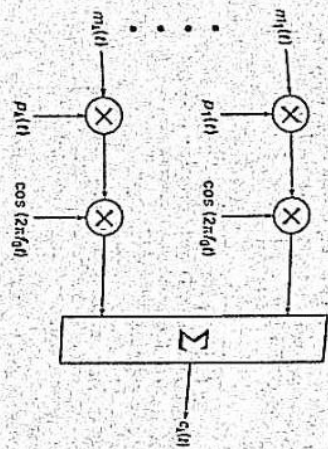


FIGURE 6.15 DS-SSMA receiver showing all the multiple channels.

Consider the case of a receiver of the first channel ($k = 1$). The output, $c_1(t)$, of the multiplier with $p_1(t)$ will be

$$c_1(t) = \sum_{i=1}^k m_i(t) p_i(t) p_1(t) s_i(t). \quad (6.12)$$

Once again, this signal will be the input to a BPSK demodulator, as shown in Figure 6.16. The input, $c_{1\text{int}}(t)$, to the demodulator can now be expressed as

$$c_{1\text{int}}(t) = \text{signal} + \text{noise} \quad (6.13)$$

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

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

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

$$p(e) = \frac{1}{2} \operatorname{erfc} \sqrt{\frac{K}{k-1}}. \quad (6.15)$$

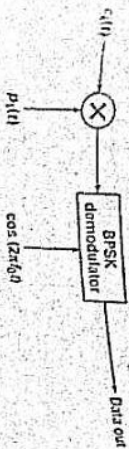


FIGURE 6.16 Demodulator.

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-SS-SSMA 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 `erfcinv()`, we can solve for $z = \text{erfcinv}(1 - 2 \times \text{BER})^2$. We get $z = 4.77$. We are given $k = 24$; $K = 23 \times 4.77 = 109.82$. Since $K = (\text{chip rate}) / (\text{data rate})$, chip rate = $109.82 \times 9600 = 1.05 \text{ Mcbps}$.

EXAMPLE 6.2

A DS-SS-SSMA 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
 Chip rate = $K \times \text{data rate} = 9,600,000 = 9.6 \text{ Mcbps}$.
 BW = $2 \times 9.6 \text{ MHz} = 19.2 \text{ MHz}$.

6.3.4 RAKE Receiver in DS-SS-SSMA 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-SS-SSMA 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 (Sklar 1993; Chan 1993a; Plick 1982, 1991; Kohn 1995; Pata 1995; Pata 1996). The fact that the PN sequences have very low correlation makes it possible to separate these delayed versions and separately perform the correlation for each block diagram in Figure 6.17 shows the concept of a RAKE receiver implementation for DS-SS-SSMA 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 precoding of the data bits.

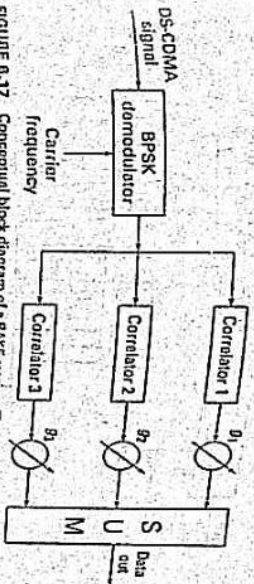


FIGURE 6.17 Conceptual block diagram of a RAKE receiver. Three multipath paths are being resolved and processed in three separate correlators. Each output is weighted by a factor g_j , $j = 1, 2, 3$.

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 ns, 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

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

The chip rate must be greater than $1/T_c = 1 \text{ Mcbps}$.

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 method uses the hopping 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 *hopper* (Sklar 1988; Taub 1986; Kohn 1995). The starting point of a FH/SS technique is either a BPSK signal or a MFSK signal. If one considers a conventional BPSK signal, the carrier frequency is alternated between two fixed frequencies, a FH/BPSK system, the data symbol modulates a carrier whose frequencies are *predefinedly* determined. This statement can be understood with the aid of the block diagram of a FH/BPSK modulator shown in Figure 6.18.

The output of a conventional BPSK modulator and the output from a digital frequency synthesizer are applied to a mixer. The bandwidth of the carrier frequency to be frequency-hopped is the product modulation. The necessary bit stream of the PN sequence drive the frequency synthesizer, enabling the carrier frequency to hop over 2ⁿ distinct values. For a given hop, the bandwidth of the bandwidth occupied by the transmitted signal will be orders of magnitude higher than for conventional BPSK. Indeed, it is possible to have a spectral spreading in FH/BPSK for exceeding that

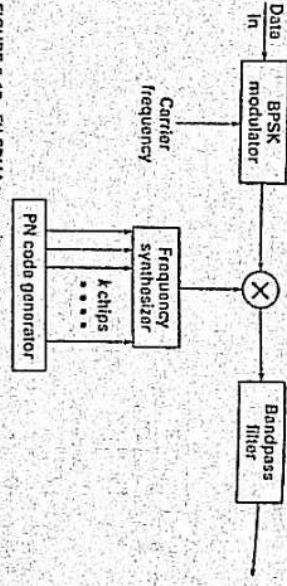


FIGURE 6.18 FH-CDMA transmitter.

observed in direct-sequence techniques. Consequently, the processing gain in FH/BPSK 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

$$R_c = \max(R_s, R_h) \tag{6.16}$$

Figure 6.19 demonstrates the difference between FFH and SFH. In Figure 6.19a, 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 6.19b, 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 BPSK receiver, the performance of FH spread-spectrum systems can be evaluated. The probability of error, $p(e)$, associated with a noncoherent BPSK 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 range of frequencies and can, likewise, be hopped over a wide bandwidth. The synchronization problem 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 compared with MHz rates

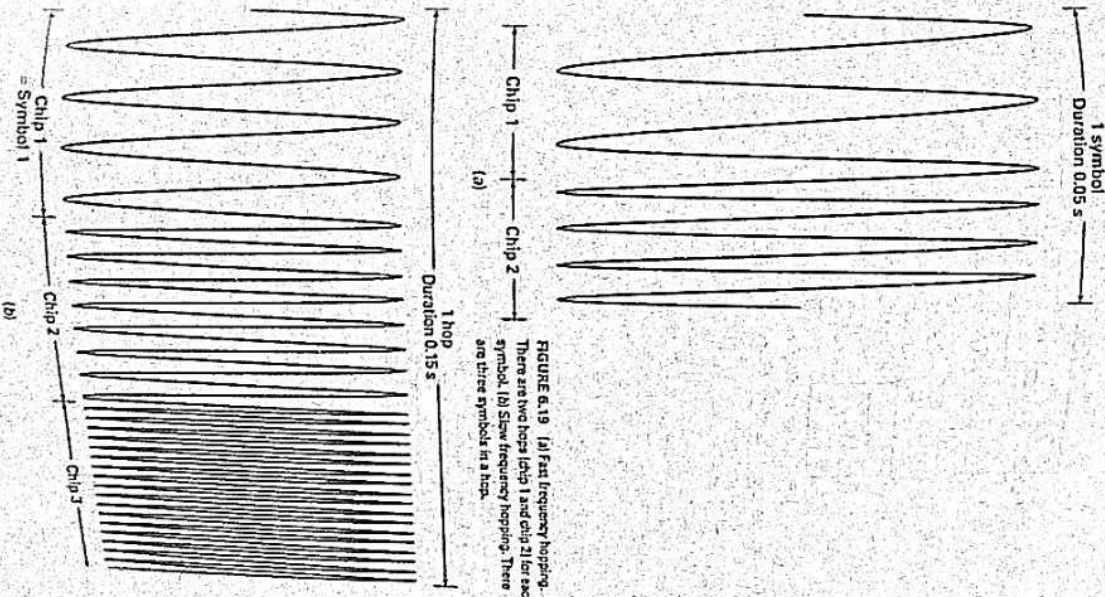


FIGURE 6.19 (a) Fast frequency hopping. There are two hops (chip 1 and chip 2) for each symbol. (b) Slow frequency hopping. There are three symbols in a hop.

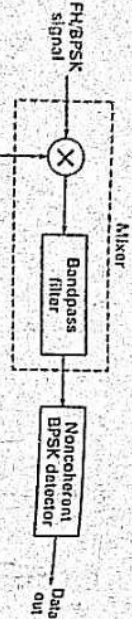


FIGURE 6.20 FH-CDMA demodulator.

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 is 869–894 MHz is used. Between the uplink and downlink communication, this large 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 ± 1 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 whenever it is. The base station reverse control

TABLE 6.1 Characteristics of AMPS

Parameter	AMPS Specification
Multiple access	FDMA
Duplex technique	FDD
Channel bandwidth	30 kHz
No. voice channels/channel	1
Uplink (reverse channel) band	824–849 MHz
Downlink (forward channel) band	869–894 MHz
Modulation technique (voice)	FM
Modulation technique (control channel)	FSK
Peak frequency deviation	± 12 kHz (FM); 20 kHz (FSK)
Data rate on control channel	10 kbps
Spectral efficiency	0.33 bps/Hz
Total number of voice channels	632

channel continuously monitors the transmission from the MU. The wideband FSK data is used in a blank-and-burst mode to initiate hand-offs to adjoining cells, change the MU transmit power, and, if necessary, initiate a hand-off to another channel in an "intra-cell hand-off mode" in places where sector antennas are used. Note that these "blank-and-burst" events are of extremely short duration (~100 ms), and any interruptions caused by them are not discernible. The typical parameters associated with AMPS are shown in Table 6.1.

To limit the peak frequency deviation to a preset value, the voice signal is passed through a compander and associated electronics, as shown in the block diagram in Figure 6.21. Since human speech has a large dynamic range, it is necessary to limit the amplitude range by compressing it in a nonlinear fashion, that is, by a 1 dB increase at the output of the compander for every 2 dB of increase in input. This confines the signal energy to the allocated bandwidth of 30 kHz. Note that at the receiver, an inverse process recovers the original speech characteristics.

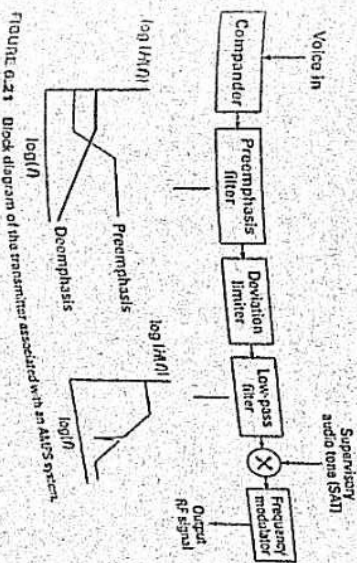


FIGURE 6.21 Block diagram of the transmitter associated with an AMPS system.

The preemphasis filter improves the performance of the FM and has the function shown in the figure. At the receiver, a deemphasis filter performs the inverse operation to preserve the signal characteristics. The low-pass filter, with a noise bandwidth of 6 MHz, serves two purposes: It limits the spectrum of the signal going into the demodulator and provides a place to insert the supervisory audio tone (SAT) signal at around 6 kHz. The SAT signal allows the BS and MU to confirm the existence of a proper link, in which it is used to indicate the "call termination" process. This tone thus acts like an on-off hook feature of a plain old telephone system (POTS).

The control data channel uses Manchester code binary frequency shift keying at a rate of 10 kbps.

Hand-off Procedure The hand-off process from one base station to another is controlled by the MTSO (mobile telephone switching office). The MTSO measures the signal strengths from the current BS and the neighboring ones. Initiation of the hand-off is based on two thresholds. For a current user, if the power is around -100 dBm, a hand-off is initiated. A threshold of -90 dBm is set for a new call.

6.4.2 United States Digital Cellular (IS 54)

The AMPS system has a limited ability to support the need for increased capacity. Digital systems offer higher capacity and the possibility of advanced features such as voice mail and paging. The U.S. digital system was designed to operate using the same frequency range, channel bandwidth, and frequency reuse plans as AMPS.

Some of the characteristics of IS 54 are given in Table 6.2. The radio-frequency (RF) channel bandwidth is 30 kHz, the same as for AMPS; however, in IS 54, three users share the bandwidth, thereby using only 10 kHz/voice channel. The gross bit rate is 48 kbps, giving each user 16.2 kbps. This individual rate is split as follows: speech = 7.95 kbps, error correction = 5.05 kbps, control channel = 0.6 kbps. This leaves 2.6 kbps as the overhead for such things as guard time, ramp-up, and synchronization, resulting in an overhead rate of 2.6/16.2 or 16%.

The modulation scheme used is $\pi/4$ -DQPSK, with a symbol rate of 24.3 kbps and a symbol duration of 41.1523 μ s. Raised cosine pulse shaping is used to reduce the effect

TABLE 6.2 Characteristics of U.S. Digital Systems (IS 54)

Parameter	Specification
Multiple access	TDMA/FDMA
Duplex	FDD
Channel bandwidth	30 kHz
Uplink (reverse channel) band	834-849 MHz
Downlink (forward channel) band	869-884 MHz
Modulation technique (voice)	$\pi/4$ -DQPSK
Forward/reverse channel data rate	48.6 kbps
Spectral efficiency	1.62 bps/Hz
Number of users/channel	3 at 235 kbps/user
Equalizer	Unspecified

of intersymbol interference. The specific demodulation method varies from manufacturer to manufacturer.

Hand-off Procedure The hand-off procedure employed in the U.S. digital system is different from the technique used in AMPS, where MTSO controls the sequence of events. In mobile-assisted hand-off (MAHO), the responsibility for the hand-off rests with the mobile unit. In this scheme, the MU measures the quality of the signal received at the mobile unit from all the base stations and continuously compares it with its base station. The MU initiates a hand-off when it determines that the power from another base station is higher than the power from its own base station by a certain amount. Thus the process is quicker than MTSO-initiated hand-off.

6.4.3 Japanese Digital System (JDC)

The Japanese digital system is remarkably similar to the American Digital Cellular (ADC) system based on IS 54. The only difference is the use of a 35 kHz RF channel bandwidth instead of the 30 kHz used in ADC. Once again, three voice channels are present in every RF channel. The modulation technique employed is $\pi/4$ -DQPSK. The gross bit rate of 42 kbps is lower than the rate for ADC, resulting in a spectral efficiency of 1.68 bps/Hz. The difference of 2.8 kbps between 14 kbps/user and 11.2 kbps/user, for protected speech, results in an overhead rate of 2.8/14 or 20%. The JDC specifications are given in Table 6.3.

6.4.4 GSM

Global System for Mobile (GSM) is a second-generation cellular system widely used throughout the world (Europe, Australia, and parts of Asia and the United States). Some specifications of GSM are given in Table 6.4. The spectral efficiency of GSM is less than the spectral efficiencies of systems based on linear modulation schemes such as the ones used in ADC and JDC.

TABLE 6.3 Characteristics of Japanese Digital Cellular

Parameter	Specification
Multiple access	TDMA/FDMA
Duplex	FDD
Channel bandwidth	25 kHz
Uplink (reverse channel) 1 band	810-826 MHz
Downlink (forward channel) 1 band	910-926 MHz
Uplink (reverse channel) 2 band	1429-1441 MHz
Downlink (forward channel) 2 band	1477-1489 MHz
Uplink (reverse channel) 3 band	1453-1465 MHz
Downlink (forward channel) 3 band	1501-1513 MHz
Modulation technique (voice)	$\pi/4$ -DQPSK
Forward/reverse channel data rate	42 kbps
Spectral efficiency	1.68 bps/Hz
Number of users/channel	3 at 87 kbps/user
Equalizer	Unspecified

TABLE 6.4 Characteristics of GSM

Parameter	Specification
Multiple access	TDMA/FDMA
Duplex	FDD
Channel bandwidth	200 kHz
Uplink (reverse channel) band	890-915 MHz
Downlink (forward channel) band	935-960 MHz
Modulation technique (voice)	GMSK (BT = 0.3)
Forward/reverse channel data rate	270.8333 kbps
Spectral efficiency	1.5 bps/Hz
Number of users/channel	8

A unique feature of GSM is the slow frequency hopping provision. In this mode of operation, the radio carrier follows a frequency hopping pattern instead of occupying the same radio frequency. This can reduce the effects of distortion present at a given radio frequency. Frequency hopping can also reduce the effects of co-channel interference.

Hand-off Procedure The hand-off procedure followed in GSM is based on MAHO. The MU performs channel quality measurements, and hand-off is initiated in a fashion very similar to the approach discussed in connection with ADC.

6.5 NORTH AMERICAN DIGITAL CELLULAR SYSTEMS BASED ON CDMA

As discussed in Section 6.3, systems based on code division multiple access can achieve the goal of increased capacity. The CDMA system based on IS 95 became operational in 1996.

There are several differences between the wireless systems based on IS 54 and the ones based on CDMA (IS 95). In the second-generation wireless systems, the forward and reverse links use a similar mode of operation in terms of the modulation technique. CDMA-based systems use different forms of modulation as well as different modes of data encoding in the two directions (Whip 1994, Pado 1994). The other fundamental difference is the use of variable-bit-rate traffic in CDMA versus the constant data rate used in ADC, JDC, and GSM. Since CDMA uses a larger bandwidth, the question of frequency reuse does not normally arise. CDMA has a reuse pattern shown in Figure 6.22a. The reuse pattern operates in the band 824-849 MHz, and the forward link operates in the band 869-894 MHz. The user data rate, which is variable, is spread at a rate of 1.25 Mc/s. For a maximum user data rate of 9.6 kbps, this provides a spreading of 128. We will now look at these and other details of the IS 95-based systems currently operating.

6.5.1 Forward CDMA Channel

The modulation format used in the forward channel is filtered QPSK. A pilot is also simultaneously transmitted so that the MU can use a coherent detection scheme for

6.5 NORTH AMERICAN DIGITAL CELLULAR SYSTEMS BASED ON CDMA 229

demodulation purposes. The pilot also provides a means of judging the signal strengths from different base stations, allowing decisions to be made on hand-off strategies.

6.5.2 Reverse CDMA Channel

Since the MU cannot afford to expend unnecessary power, the pilot signal is not transmitted in the reverse direction. The modulation format used in the reverse link is DQPSK.

6.5.3 Power Control in CDMA Systems

Since all the terminals (MUs) transmit over the same frequency band, power control is essential in the reverse channel. At the base station, the signal from the desired user is likely to be swamped by other users who are "nearby." This receiver problem is caused by the absence of complete decorrelation between the codes of different users. The problem can be alleviated through power control, which will ensure that all the terminals within a certain geographical region arrive at the base station with almost equal power. Note that the capacity of a CDMA-based system is limited by the interference. This interference can be reduced and capacity enhanced by using power control. Power control also plays a role in conserving transmitted signal power, thereby increasing the battery recharge cycle.

Power control is usually performed in two ways: using an "open-loop" method or using a "closed-loop" method. In the former approach, it is assumed that transmission characteristics are identical in the forward and reverse directions, and therefore the power loss should be the same in the two directions. The terminal keeps the total power, the sum of the transmitted and received power, at a constant level (-73 dBm). The constant monitoring of the received power allows control of the transmitted power from the terminal. In the closed-loop method, the base station monitors the power of the terminal and the commands to adjust the power in steps of 21 dB at a very fast rate (~800 times/s).

Note also that IS 95 systems use voice-activated transmission. This means that a carrier signal is transmitted only when a voice signal is detected. This reduces overall interference by the ratio of this "silence time" to the whole speech period.

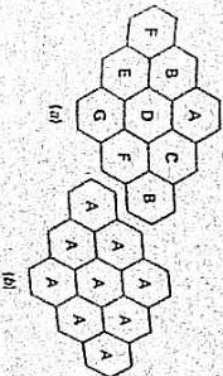


FIGURE 6.22 (a) Spread-code reuse pattern; (b) CDMA reuse pattern.

6.5.4 Hand-off Procedure

The hand-off procedure in IS 95 systems is unique and is referred to as "soft hand-off" (Wong 1997, Gang 1999). As the mobile unit moves from cell to cell, it communicates simultaneously with the two base stations (Figure 6.23).

Essentially, there is no break in the transition from one base station to the next—hence the term *soft hand-off*. This not only makes the switch unnoticeable, but also reduces the chances of being dropped, distinguishing it from the MAHO technique employed in second-generation digital systems.

After initiation of the call, the terminal continues to monitor whether the signal from another cell is comparable to the signal from its own base station. If the signal is strong, the terminal informs the MTSO that the new cell site is strong and identifies the new site. Based on this, the MTSO starts the hand-off by linking to the new cell site while maintaining the connection to the existing base station. Thus for a short period, the terminal achieves a sense of diversity by having two base stations simultaneously communicating with it. This simultaneous connection with the two base stations also eliminates the "ping-pong" effect, which is a result of repeated requests to transfer back and forth between two base stations—a common occurrence in systems not having this unique "diversity."

6.5.5 Diversity in CDMA Systems

The high bandwidth of the signal (~1.25 MHz) provides a unique advantage in terms of time diversity. The chips are so narrow that multipath fading, which results in overlapping, unresolved pulses in TDMA/FDMA-based systems, produces nonoverlapping multiple pulses, resulting in time diversity. A RAKE receiver can therefore be used to combine these multiple versions of the signal.

Unique Features of IS 95

- Frequency reuse of unity
- Automatic power control
- Time diversity through a RAKE receiver
- Soft hand-off
- Voice-activated transmission

A detailed discussion of the capacity of CDMA-based systems is given in Section 6.6.1.

6.6 COMPARISON OF MULTIPLE-ACCESS SYSTEMS IN WIRELESS COMMUNICATIONS

One parameter used for comparison of the various multiple-access systems is the channel capacity (Lee 1991a, Rahn 1991, Glin 1993, Kohn 1993). Channel capacity, or radio capacity, is a quantitative measure of the ability of the access scheme to provide a maximum number of channels in a given bandwidth. As we saw in Chapter 4, we can certainly pack more channels into a given bandwidth if we ignore the effect of co-channel interference (CCI). Interference from cells operating at the same carrier frequency will degrade the performance of the communication system.

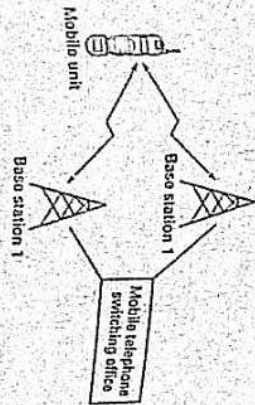


FIGURE 6.23 Hand-off procedure in CDMA system.

If R is the radius of the cell and D is the distance from the mobile unit to the "interfering channel," or the base station responsible for the interfering channel, the co-channel interference reduction factor or frequency reuse factor is given by eq. (4.8),

$$q = \frac{D}{R} \tag{6.18}$$

The signal-to-CCI ratio, C/I , is expressed as

$$\frac{C}{I} = \frac{R^{-\alpha}}{I} = \left(\frac{1}{6} \right) \left(\frac{R}{D} \right)^{-\alpha} \tag{6.19}$$

where it is assumed that all the interfering stations are separated from the desired channel by the same distance D and have the same loss exponent, α . In any practical system, the actual C/I must be greater than some acceptable value of $(C/I)_{\text{acc}}$. Using eqs. (4.7) and (4.8), the expression for the co-channel interference reduction factor, q , can be rewritten as

$$q = \left[6 \left(\frac{C}{I} \right)_{\text{acc}} \right]^{1/\alpha} \tag{6.20}$$

The radio capacity of the system, m , is defined as

$$m = \frac{\text{total allocated spectrum } (B_t)}{\text{channel bandwidth } (B_c)} \times \text{number of cells } (N) \tag{6.21}$$

Making use of the relationship between q and N ,

$$q = \sqrt[3]{N}, \tag{6.22}$$

the expression for the radio capacity becomes

$$m = \frac{3B_t}{B_c} \left[6 \left(\frac{C}{I} \right)_{\text{acc}} \right]^{3/\alpha} \tag{6.23}$$

Using a value of $\nu = 4$, the expression for the capacity becomes

$$m = \frac{3B_i}{B_i \sqrt{\frac{2}{3}} \left(\frac{C}{I}\right)_{\min}} \quad \text{radio channels/cell} \quad (6.24)$$

If we wish to achieve the same capacity for two different types of systems, we can write

$$B_i \nu_i \sqrt{\left(\frac{C}{I}\right)_{\min i}} = B_i \nu_j \sqrt{\left(\frac{C}{I}\right)_{\min j}} \quad (6.25)$$

where i, j correspond to the two systems. In other words, if we have a narrowband system that operates with a lower bandwidth than another system, the signal-to-CCI ratio must be high. Conversely, if we have a constant number of users/radio channel, $(C/I)_{\min}$ increases by 4 when the bandwidth is halved.

We now compare the capacity of analog mobile systems and digital mobile systems. Even though the digital systems use less bandwidth, because of the increased value of $(C/I)_{\min}$, the capacities are the same. However, in practice, the capacities of digital systems are higher than those of analog systems because the use of spread and error-corrective coding permits operation with lower $(C/I)_{\min}$ values. Table 6.5 illustrates this aspect of capacity improvement in digital mobile systems. It has been assumed that each cell site has a 120° antenna. A GOS of 25% has been used. With the use of mobile-assisted hand-off (MAHO), the monitoring of the base stations becomes easier and the best station choice can easily be made. Indeed, MAHO allows the use of densely packed microcells, which tends to increase capacity.

6.6.1 Capacity of CDMA Systems

The capacity of TDMA and FDMA systems depends on the available bandwidth and the CCI ratio. The capacity in CDMA systems depends solely on the interference from the other users since the channel is shared simultaneously by all users (Jung

TABLE 6.5 Capacity Gains of Digital Systems over AMPS

Parameter	Cellular system			
	AMPS	USDC	JDC	GSMT
Access method	FDMA	TDMA	TDMA	TDMA
Bandwidth (MHz), B_i	25	25	25	25
Carrier spacing bandwidth (kHz), B_c	30	30	25	200
User/carryer	1	3	3	8
Total number of voice channels	833	2600	3000	1000
Frequency reuse (cluster size)	7	7	7	4
Channels/site	119	357	429	250
Traffic (Erlangs) ^a	11.9	41	50	217
Capacity gain over AMPS	1	3.45	4.20	2.30

^a The per-cell traffic has been assumed to be 1 Erl. The capacity of a hexagonal cell area is $2.6A \sqrt{3} \rho$, where A is the area of the cell and ρ is the traffic density. For a 120° antenna, $A = 0.173 \times 10^6 \text{ m}^2 = 173,000 \text{ m}^2$. For a 120° antenna, $\rho = 10 \text{ Erl/km}^2$, giving the total capacity of 4638, 1574, 120, and 115 Erl, respectively. The traffic value given is obtained by dividing the total capacity by the number of cells.

1993, Kim 1993, Paulo 1994). Thus, if we can decrease the interference from other users, the capacity of a CDMA system is likely to increase.

Let us start by considering a single cell with N users who share the cell. The interference to any one of the users is a result of the $(N - 1)$ other users sharing the bandwidth of the cell. If we have power control, all the terminals will be transmitting with the same power. If S is the signal power from the desired terminal, the signal-to-noise ratio due to interference can be expressed as

$$\text{SNR} = \frac{S}{(N-1)S} = \frac{1}{N-1} \quad (6.26)$$

Instead of looking only at the signal-to-interference ratio, we can look at the signal-to-noise ratio in terms of the energy in the bit. The signal-to-noise ratio at the base station receiver, given by E/N_0 , can be expressed as

$$\frac{E}{N_0} = \frac{S/R}{(N-1)S/R} = \frac{B_i}{R(N-1)} \quad (6.27)$$

where R is the information bit rate and B_i is the RF bandwidth. We have divided the signal power (S) in the numerator by the bandwidth (data rate R) of the message data, while the signal power in the denominator has been divided by the bandwidth (B_i) occupied by the interfering signal. The quantity (B_i/R) is the processing gain K of the CDMA processing, defined in connection with Figure 6.7.

Note that eq. (6.27) has assumed that the reception is influenced only by the interference from other users. However, in almost all systems, additional degradation results from the presence of thermal noise in the system. Including thermal noise of total power over the whole available bandwidth, η_b , the equation for the signal-to-noise ratio (6.27) can be expressed as

$$\frac{E}{N_0} = \frac{K}{(N-1) + \eta_b/S} \quad (6.28)$$

At this point, let us understand the meaning of eq. (6.28). By including both the thermal noise and the interference noise, the left-hand side can be interpreted as the effective signal-to-noise ratio observed at the receiver. We can now explore ways of fixing the maximum number of users that can be supported in the system if we expect to maintain an acceptable level of performance. This acceptable level will come from the acceptable value of the bit error rate, which in turn will dictate a minimum acceptable value of the signal-to-noise ratio. If the minimum signal-to-noise ratio required to maintain an acceptable error rate is $(E/N_0)_{\min}$, the maximum number of users N_{\max} that can be supported will be given by

$$\left(\frac{E}{N_0}\right)_{\min} = K \left(N_{\max} - 1 + \frac{\eta_b}{S}\right)^{-1} \quad (6.29)$$

However, for us to use eq. (6.28b), we need to convert η_b to a more meaningful quantity.

If $N_{th}/2$ is the spectral density of the thermal noise, we can write

$$\eta_b = B_{th} N_{th} \quad (6.30)$$

We can rewrite eq. (6.28b) using eq. (6.29) as

$$\left(\frac{E}{N_0}\right)_{\min} = K \left(N_{\max} - 1 + \frac{B_{th} N_{th}}{S}\right)^{-1}$$

We can now express eq. (6.30) in terms of the signal-to-noise ratio of the message signal for the single-user case. The signal-to-noise ratio (energy) can be expressed as S/N_0 . This is the signal-to-noise ratio if only a single signal is to be transmitted, and we will identify this as $(E/N_0)_s$. We can rewrite eq. (6.30) as

$$\left(\frac{E}{N_0} \right)_{\text{min}} = K \left(N_{\text{max}} - 1 + K \left(\frac{E}{N_0} \right)_s^{-1} \right)^{-1} \quad (6.31)$$

Inverting to get the maximum number of users that can access the system with an acceptable level of performance, we get

$$N_{\text{max}} = 1 + K \left[\frac{1}{(E/N_0)_s} - \frac{1}{(E/N_0)_s} \right] \quad (6.32)$$

In other words,

$$N_{\text{max}} = 1 + K \left[\frac{(E/N_0)_s - (E/N_0)_{\text{min}}}{(E/N_0)_{\text{min}} (E/N_0)_s} \right] \quad (6.33)$$

If thermal noise is negligible, $(E/N_0)_s$ is very high, and the maximum number of users will be determined by the minimum acceptable performance level set by $(E/N_0)_{\text{min}}$. Otherwise, in the presence of thermal noise, the number of users will be reduced by a factor determined by the amount of noise present in the system.

EXAMPLE 6.4

In a DS-SSMA system, an input data stream is coming in at the rate of 2 kbits/s. If the chip rate is 200 kbits/s, what is the processing gain?

Answer The processing gain is given by the ratio of the chip rate to the data rate:

$$K = 200/2 = 100, \text{ or } 20 \text{ dB.}$$

EXAMPLE 6.5

If there is negligible thermal noise in a system, what is the signal-to-noise ratio at the receiver? The number of users is 11. If the transmit power is increased, how will the signal-to-noise ratio be affected?

Answer We have seen that the signal-to-noise ratio in the absence of receiver noise is given by eq. (6.27). The SNR ratio is

$$\frac{E}{N_0} = \frac{K}{N-1} = \frac{100}{10} = 10, \text{ or } 10 \text{ dB.}$$

When the transmit power is increased, the signal-to-noise ratio given in eq. (6.27) is unaffected (still 10 dB) because the ratio is dependent only on the processing gain and the number of users (both unaffected by the transmit power).

EXAMPLE 6.6

In the DS-SSMA system, the incoming data rate is 10 kbits/s. The chip rate is 100 kbits/s. It has been determined that for acceptable performance, the signal-to-noise ratio $(E/N_0)_s$ at the receiver must be greater than 10 dB. If only a single signal is transmitted, the signal-to-noise ratio will be 10 dB.

- (a) What is the maximum number of users that can be supported under these conditions?
(b) Under ideal conditions, what is the maximum number of users that can be supported?

Answer

(a) We are given that the minimum value of SNR is 10 dB, or 10 in absolute units, i.e., $(E/N_0)_{\text{min}} = 10$. We are also given that signal-to-thermal noise ratio is 16 dB, or 100 in absolute units, i.e., $(E/N_0)_s = 100$. Note also that the processing gain $K = (10^3/10^2) = 100$. Making use of eq. (6.33), the maximum number of users that can be supported is given by

$$N_{\text{max}} = 1 + 1000 \left(\frac{40 - 10}{40 \times 10} \right) = 1 + 75 = 76.$$

(b) Under ideal conditions, $(E/N_0)_s$ is very high, and we make use of eq. (6.27), where the left-hand side now is the minimum acceptable signal-to-noise ratio. We see that

$$10 = \frac{1000}{N-1}$$

or $N_{\text{max}} = 101$. We can clearly see the advantage of eliminating receiver noise. The maximum number of users goes up significantly when the system noise is eliminated.

EXAMPLE 6.7

In Example 6.6, if the transmit power is reduced by 2 dB, what is the maximum number of users that will be able to share the spectrum?

Answer In an ideal scenario (i.e., if we neglect thermal noise), there will be no change in the number of users (101). However, under realistic conditions, reduction in the signal-to-noise ratio will result when the transmit power is reduced because the thermal noise remains constant. We are given $(E/N_0)_s = 14$ dB or about 25 in absolute units. Making use of eq. (6.33), the maximum number of users when the power is reduced by 2 dB is given by

$$N_{\text{max}} = 1 + 1000 \left(\frac{25 - 10}{250} \right) = 61.$$

We see that we can support 15 fewer users when the transmit power is reduced by 2 dB.

We can certainly enhance the capacity if the signal-to-noise ratio is high. We see from eq. (6.33) that such an increase in capacity can be accomplished by reducing the number of interfering channels. Going from an omnidirectional antenna to a 120° sector antenna automatically brings down the number of interfering cells to one-third of the number in the omnidirectional antenna case, thereby increasing the capacity.

A second means of increasing the capacity is to incorporate a voice activity monitor and turn off the transmitter when speech is stopped. When we speak, we normally use pauses between words, and by turning off the transmitter when no voice activity is detected, we can reduce the effective number of interfering users. If we designate α as the voice activity factor, eq. (6.27) and the equations that follow can be rewritten. Specifically, eq. (6.28) becomes

$$\left(\frac{E}{N_0} \right)_{\text{min}} = \frac{K}{(N-1)\alpha + N_0/S} \quad (6.34)$$

Typical values of α are around 3/8, and such values can bring down the effective number of interfering cells and increase the actual number of users.

It must be noted, however, that the capacity of a CDMA system is soft while the capacity of TDMA or FDMA systems is hard. In other words, CDMA will allow more users into the system at the cost of reduced quality, while TDMA/FDMA-based systems cannot allow any more users when no channels are available.

The capacity calculations presented here have ignored the fundamental differences between reverse-link and forward-link performance of CDMA. In the forward link (downlink), the MU has the availability of a pilot signal, making coherent demodulation possible. The reverse link, on the other hand, does not have the availability of a pilot tone to conserve the power of the MU. Thus, the capacity of the CDMA system is likely to be limited by the weaker of the two links, namely, the reverse link. We will derive approximate expressions for the capacity of the reverse-link CDMA, including multiple cells, the effects of which were not considered in arriving at eq. (6.33) or (6.34).

In CDMA systems, the reuse factor is unity and, hence, the same frequency is used in all the cells. This means that there will be interference occurring from users in other cells in addition to the users from the single cell under consideration. If we include an intercell interference factor f (Guth 1991, Jung 1993) in eq. (6.34), we can rewrite the equation as

$$\frac{E}{N_0} = \frac{K}{N_0} \cdot \frac{(N-1)\alpha(1+f) + \eta_0/S}{\alpha(1+f)} \quad (6.35)$$

Equation (6.35) also includes the voice activity reduction factor α . The intercell interference factor f depends on a number of factors, such as the geometry of the serving cell and the neighboring cells, the path loss factor η , the standard deviation of lognormal fading (σ_{η}), and hand-off procedures used. Estimating the intercell factor f is also a bit difficult because of the lack of control on the part of the serving cell on the users in other cells. Still, a few assumptions can be made. The intercell interference factor f will be less if the path loss factor or exponent is high. It will also be less if the size of the cell is large. Small values of the standard deviation of fading also reduce the value of f . In addition, as stated above, it is influenced by the hand-off procedures employed. It may be in the range of 0.44 to 2. It is now possible to rewrite eq. (6.31) taking this additional interference into consideration:

$$\left(\frac{E}{N_0}\right)_{\min} = K \left[N_{\text{cell}} - 1 \right] \alpha(1+f) + K \left(\frac{E}{N_0}\right)_{\min} \quad (6.36)$$

where N_{cell} is the maximum number of users in a cell, taking intercell interference into account. Solving for N_{cell} , we get

$$N_{\text{cell}} = 1 + K \left[\frac{(E/N_0)_{\min} - (E/N_0)_{\text{req}}}{(E/N_0)_{\text{req}}} \right] \frac{1}{\alpha(1+f)} \quad (6.37)$$

If we define $N_{U,0}$ to be the maximum number of users/cell in the absence of thermal noise ($(E/N_0)_{\text{req}} \gg (E/N_0)_{\text{th}}$) and intercell interference ($\alpha = 1, f = 0$), we obtain (from eq. (6.37)),

$$N_{U,0} = 1 + K \left(\frac{E}{N_0}\right)_{\min}^{-1} \quad (6.38)$$

Rewriting eq. (6.37) using eq. (6.38), we get

$$N_{\text{cell}} = N_{U,0} \left[\frac{(E/N_0)_{\min} - (E/N_0)_{\text{req}}}{(E/N_0)_{\text{req}}} \right] \frac{1}{\alpha(1+f)} \quad (6.39)$$

where we have assumed that $N_{U,0} = K(E/N_0)_{\min}^{-1}$. This is a valid assumption in most cases because $N_{U,0} \gg 1$.

If we define cell utilization N_u as the ratio of N_{cell} to $N_{U,0}$, we can write

$$N_u = \left[\frac{(E/N_0)_{\min} - (E/N_0)_{\text{req}}}{(E/N_0)_{\text{req}}} \right] \frac{1}{\alpha(1+f)} \quad (6.40)$$

The limiting value of N_u in the absence of voice activity reduction ($\alpha = 1$) is realized when $(E/N_0)_{\text{req}}$ goes to ∞ and is given by

$$N_{u,\max} = \frac{1}{(1+f)} \quad (6.41)$$

Thus, the ability of the CDMA system to provide access to a large number of users is limited by the interference introduced by the neighboring cells.

Equation (6.40) gives us a measure of the performance of the CDMA system in terms of the required signal-to-noise ratio $(E/N_0)_{\text{req}}$, the signal-to-thermal noise ratio $(E/N_0)_{\text{th}}$, the voice activity factor α , and the intercell interference factor f . Let us now explore this dependence of the capacity of the CDMA system on these various parameters.

EXAMPLE 6.8

In a DS-SS-CDMA system, the minimum signal-to-noise ratio $(E/N_0)_{\text{req}}$ for acceptable performance is 12 dB. For a single channel, the signal-to-noise ratio $(E/N_0)_{\text{th}}$ is 18 dB. If the processing gain is 10 MHz/s and the chip rate is 10 Mbps, calculate the maximum number of users supported. If an intercell interference parameter of 1.2 is observed, calculate the maximum number of users. Assume that no voice reduction factor is included in the calculation.

Answer

$$K = 1000$$

$$\left(\frac{E}{N_0}\right)_{\text{req}} = 18 \text{ dB, or } 63 \text{ in absolute units}$$

$$\left(\frac{E}{N_0}\right)_{\text{th}} = 12 \text{ dB, or } 15.85 \text{ in absolute units}$$

Using eq. (6.33),

$$N_{\text{cell}} = 1 + 1000 \times \left[\frac{63 - 15.85}{15.85} \right] = 148$$

Including the intercell interference factor, in terms of eq. (6.37),

$$N_{\text{cell}} = 1 + 1000 \times \left[\frac{63 - 15.85}{15.85} \right] \frac{1}{(1+1.2)} = 21$$

Intercell interference reduces the number of users by more than 50%. ■

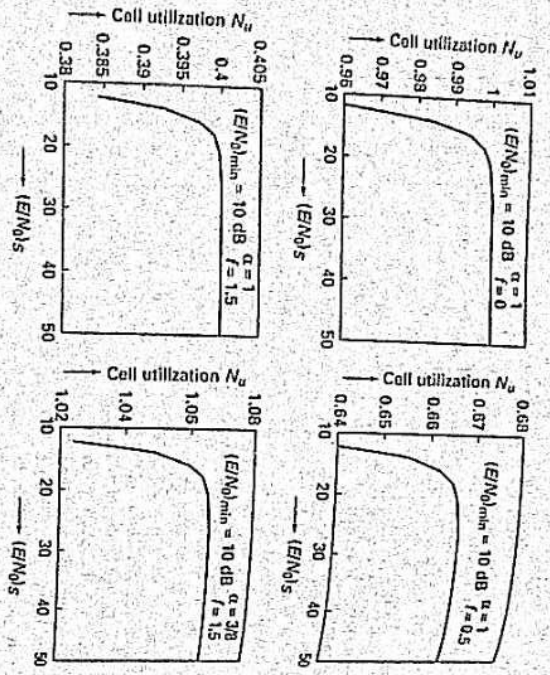


FIGURE 6.24 Plots of the cell utilization N_u for different parameters.

The cell utilization parameter N_u is plotted in Figure 6.24 for different conditions in the geographical area. It is clear that for intercell interference of 0.5, the utilization is only in the 60% range. It goes down to the 40% range when the interference factor goes up to 1.5. Once we incorporate the voice activity reduction factor, we see that the utilization exceeds 100%, pointing to the possibility of accommodating more users even in the presence of intercell interference.

A few statements are in order here. From Example 6.5, we see that an increase in thermal noise and an increase in intercell interference lead to a reduction in the number of users. This does not imply that there is a cap on the number of users. Indeed, when more users come into the system, nobody is turned away, and the performance level goes down. In other words, $(E/N_0)_{min}$ will no longer be the minimum acceptable value. It will be a floating value that keeps going down as the number of users entering the system goes up.

Let us explore this further. If acceptable performance is defined by $(E/N_0)_{min}$, the maximum number of users is given by $N_{u,0}$ given in eq. (6.38). We will also assume that there is no intercell interference. Rewriting eq. (6.32), we get

$$\frac{N-1}{K} + \left(\frac{E}{N_0}\right)^{-1} = \left(\frac{E}{N_0}\right)^{-1} \quad (6.42)$$

We can now obtain an expression for the effective signal-to-noise ratio $(E/N_0)_{eff}$ as the number of users N increases:

$$\left(\frac{E}{N_0}\right)_{eff} = \frac{K}{N-1 + K(E/N_0)^{-1}} \quad (6.43)$$

We will look at this reduction in signal-to-noise ratio as more and more users come into the system (see Figure 6.25). If the number of users in the system is the maximum allowed (for the case in this figure, $N_{max} = 10$), the minimum acceptable signal-to-noise ratio is chosen to be 10 dB. As additional users enter the system, the actual or effective signal-to-noise ratio goes down.

The results for two values of the signal-to-thermal noise level $(E/N_0)_0$ are also shown. At a very low noise value (high value of the ratio = 50 dB), the SNR values are better than those at a high value of thermal noise (low value of the ratio = 18 dB). These results show that the introduction of additional users will reduce the performance level, though the level may still be good enough for use of the wireless system.

Figure 6.26 shows the improvement in signal-to-noise ratio at the receiver when the actual number of users is less than the maximum number based on acceptable performance. We see that when there are fewer users, it is possible to have a higher-than-required signal-to-noise ratio in the receiver. The effect of reducing thermal noise as seen here is similar to the one seen in Figure 6.25.

6.6.2 Comparison of Features and Standards

Table 6.6 compares some of the features of multiple-access schemes, and Table 6.7 compares the various standards on the basis of several parameters.

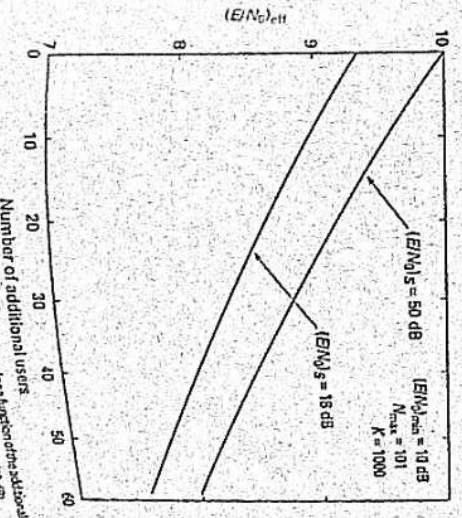


FIGURE 6.25 The effective signal-to-noise ratio as a function of the additional users beyond the maximum set by the acceptable SNR value at the receiver (10 dB).

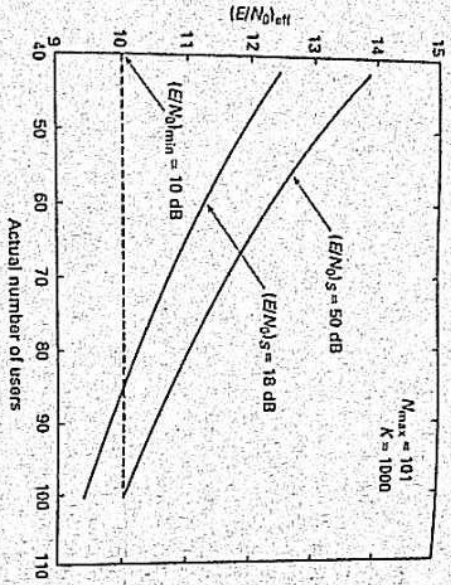


FIGURE 6.26 Enhancement in signal-to-noise ratio at the receiver as a function of the number of users in the cell.

6.7 CDMA2000

The multifold increase in the use of wireless services (both voice and data) has made it necessary to increase the capacity of the wireless communication systems. Combined with this need for increased capacity, there has been an interest in providing a greater variety of services, such as video-on-demand and other multimedia services (Alysa 1997; Kris 1998a,b; Rao 1999; Gang 1999; Prus 1999). The goals of increased capacity, diversity of applications, and universal use of wireless systems are to be achieved through the IMT 2000 (International Mobile Telecommunications 2000) initiative. The major goals of IMT 2000 are the universal adoption of a single standard and global roaming capability, with deployability indoors, outdoors, in mobile or stationary environments.

TABLE 6.0 Features of FDMS, TDMA, and CDMA

Feature	FDMA	TDMA	CDMA
High carrier frequency stability	Required	Not necessary	Not necessary
Timing synchronization	Not required	Required	Required
Power monitoring	Difficult	Easy	Easy
Receiver problem	No	No	Yes: power control required
Variable transmission rate	Difficult	Easy	Easy
Fading mitigation	Equalizer not needed	Equalizer may be required	RAKE receiver possible
Zone size (large/small)	Any size	Any size	Large size difficult because of power control

TABLE 6.7 Comparison of Standards

Parameter	IS 54	JDC	IS 95
Multiple access	TDMA/FDMA	TDMA/FDMA	TDMA/FDMA
Duplex	FDD	FDD	FDD
Channel bandwidth	30 MHz	25 MHz	20 MHz
Uplink (reverse) channel 11 band	824-849 MHz	810-876 MHz	824-849 MHz
Downlink (forward) channel 11 band	869-894 MHz	940-959 MHz	814-834 MHz
Modulation technique (voice)	$\pi/4$ DQPSK	$\pi/4$ DQPSK	QPSK
Forward/reverse channel data rate	48.6 kbps	42 kbps	270 kbps
Forward/reverse channel symbol rate	24.3 ksymb/s	21 ksymb/s	270 ksymb/s
Users/channel	3 at 7.95 kbps/user	3 at 6.7 kbps/user	6 at 13 kbps/user
Number of voice channels	2500	3000	1000
Spectral efficiency	1.62 bps/MHz	1.68 bps/MHz	1.35 bps/MHz
Equalizer	Adaptive	Adaptive	Adaptive

in megacells (i.e., global cells) (> 35 km radius), in macrocells (1-35 km radius), in microcells (up to 1 km radius), and in picocells (a few meters radius). The planned scope of IMT 2000 is sketched in Figure 6.27 (Shaf 1998; Shum 1998; Kris 1998b).

The third-generation (3G) systems based on IMT 2000/UTMS (Universal Mobile Telecommunication Systems) are being put in place in Europe, Asia, and other parts of the world (Sasa 1998; Sami 1998; Wei 1998; Rao 1999). The North American system designated as cdma2000 can support all the IMT 2000 requirements as well as 2G systems enhancements. The cdma2000 radio transmission technology (RTT) is a wideband spread-spectrum-based interface using the code division multiple access (CDMA) approach. It also provides backward compatibility with IS 95 CDMA systems currently in operation. While a new spectral band around 1900 MHz is being used for these systems in most of the world, the PCS band of 1900 MHz, which is currently used for the second-generation digital systems, is being used in North America for cdma2000.

Some of the unique features of cdma2000 are as follows:

- Compatibility with IS 95
- Possibility of multiple bandwidths: 1.25 MHz, 3.75 MHz, 7.5 MHz, 11.25 MHz, and 15 MHz in multiples of 1.3, 6, 9, and 12 of 1.25 MHz
- Chip rates of $N \times 1.2228$ Mcbps, with $N = 1, 3, 6, 9, 12$
- Multicarrier transmission in the forward (down) link, making frequency diversity possible
- Pilot-based coherent detection in both forward and reverse links

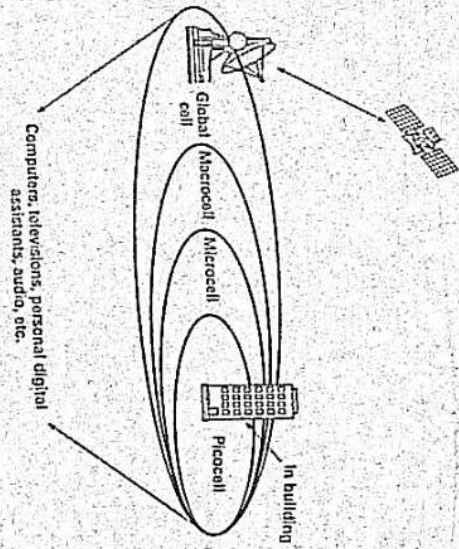


FIGURE 6.27 Vision and scope of IMT 2000.

We will now review some of these characteristics and compare them with those of W-CDMA, the prime candidate for IMT 2000 implementation in Europe.

Before we can compare the features of cdma2000 and W-CDMA, we need to look at the similarities between the existing CDMA-based systems operating in North America, namely, the features associated with IS 95 and cdma2000. Keeping in mind the need to provide a bridge between IS 95 and cdma2000, modifications to the original IS 95 (known as IS 95A) were made and designated as IS 95B. This modified IS 95B supports user data rates much higher than the ones offered by IS 95A (14.4 kbps). These higher data rates are 9.6–76.8 kbps in rate set 1 and 14.4–15.2 kbps in rate set 2. The updates have been provided without changing the physical layer associated with IS 95A. These changes have made compatibility with cdma2000 possible. Table 6.8 provides the details of this comparison.

Let us briefly look at two unique features of cdma2000 (Riso 1999; Pan 1999; Gang 1999, 2000). One of these features is the ability to use direct spreading or multicarrier transmission with the availability of bandwidths of 5 MHz or more. The concept for forward link transmission is shown in Figure 6.28 for two spreading options. In the multicarrier approach (a form of orthogonal frequency multiplexing), three or more carriers are used, each with a spreading rate of 1.2288 Mcbps/s. Figure 6.28 shows such an approach with three carriers. It also shows the second approach, in which the whole bandwidth is used for direct spreading at a rate of $N \times 1.2288$ Mcbps/s. The diagram shows spreading at a rate of 3.6864 Mcbps/s, three times the spreading of one of the channels in the multicarrier approach. Use of such multicarriers is equivalent to the frequency diversity. The direct spreading over the whole bandwidth also provides the equivalent of frequency diversity.

TABLE 6.8 Comparison of IS 95 and cdma2000

Feature	IS 95 (A & B)	cdma2000
RF channel width (kHz)	125	1.25
Chip rate (Mcps/s)	1.2288	11.5 (up to 30)
Chip rate (Mbps/s)	9.6–115.2	12.96–129.6 (up to 115.2)
Single-user data rate (kbps)		9.6–29.7
Modulation	BPSK with quadrature spreading	QPSK with quadrature spreading
Pilot-based coherent detection	Forward link: yes Reverse link: no	Forward link: yes Reverse link: yes
Channel coding	Convolutional code	Convolutional code
FSS forward power control	No	Yes
FSS reverse power control	Yes	Yes
Forward link transmit diversity	No	Yes for B/W of 5 MHz or more
Use of turbo codes to lower operating SNR levels	No	Yes

The second feature is the inherent diversity associated with the use of multicarriers (see Appendix B, Section B.1). These multiple carriers can be transmitted using a technique known as a multi-antenna. Two multi-antenna configurations are shown in Figure 6.29.

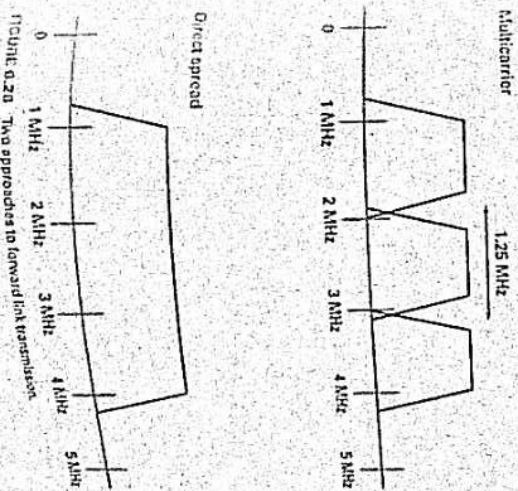


FIGURE 6.29 Two approaches to forward link transmission.

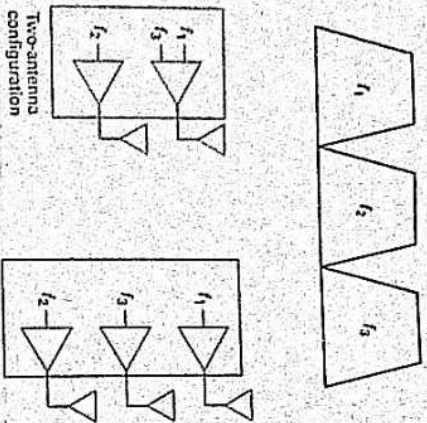


FIGURE 6.29 Transmit diversity configuration for the forward link ($M = 3$).

In one of the configurations, two nonconsecutive frequencies, f_1 and f_3 , are transmitted on one antenna while f_2 is transmitted on a second antenna. In the second configuration, three separate antennae are used to transmit the three frequencies f_1 , f_2 , and f_3 . Use of diversity allows cdma2000 to mitigate fading effects and therefore improves the performance, leading to increased forward link capacity.

We can now compare the two schemes cdma2000 and W-CDMA (Ruka 1996; Kralj 1998a,b). The two share a number of unique features that do not exist in the current IS 95 and GSM-based systems. Some of the important common features are the following:

- Coherent forward and reverse links
- Fast forward link power control and fast reverse link power control
- Turbo codes for higher rates
- Variable spreading factor to achieve higher data rates
- QPSK modulation on both forward and reverse links

There are also a few key differences between the two systems. They are listed in Table 6.3.

TABLE 6.3 Comparison of cdma2000 and W-CDMA

Feature	cdma2000	W-CDMA
Chip rate	3.6864 Mcbps	4.096 Mcbps
Network support	IS 41	GSM, MAP (mobile application part)
Base station synchronization	Synchronous	Asynchronous
Multicarrier spreading	Yes	No

6.8 OTHER NEW DEVELOPMENTS: BLUETOOTH NETWORKS

With the increasing use of computers, printers, fax machines, cell phones, and PDA (personal digital assistants), it becomes necessary to exchange data among these units without wires or cables, as shown in Figure 6.30. Networks based on infrared (IR) systems are one option to accomplish this goal. However, the use of such systems is limited due to their high cost and line-of-sight and directional characteristics. Because of the necessity for links that operate smoothly and efficiently, these networks must possess the following unique features:

- The network must operate globally. This means that the frequency band must be available globally. It must also be license free and open to all radio systems.
- The network connections can be made on an ad hoc basis. Anyone should be able to bring in any unit and achieve connectivity.
- The network must support voice and data.
- The radio transceiver must be small and operate at a low power level. Such operation will keep the interference low. Low interference is a must because license-free frequency bands are likely to be used anywhere, anytime. The typical range will be from 0.1 to 10 m.

Bluetooth was developed by Ericsson to meet these goals (Gibb 2000, Herz 2000, Schin 2000, Zurb 2000). Bluetooth-enabled products will automatically set each other and configure themselves into networks. We will briefly look at some of the characteristic features of the Bluetooth wireless network system.

The frequency range that meets the requirement is the ISM (industrial, scientific, medical) band in the 2.4 GHz range. Since the ISM band is open to anyone, systems operating at this band must cope with several sources of interference from such units as baby monitors, garage door openers, cordless phones, and microwave ovens. To overcome the interference, the frequency-hopping spread-spectrum format is used. The modulation format used is Gaussian-shaped frequency shift keying (GFSK).

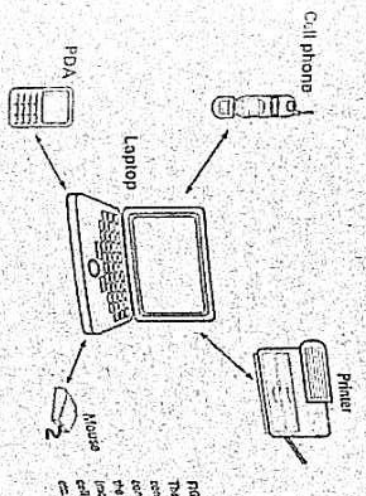


FIGURE 6.30 The Bluetooth concept. It uses connectivity of IR technology to link peripheral devices (laptop, printer, mouse, PDA, and phone) and phone PDA and phone PDA and phone PDA and phone PDA.

TABLE 6.10 Characteristic Features of Bluetooth Wireless Networks

Carrier frequency (MHz)	2400-2483.5 (ISM band)
Modulation	<ul style="list-style-type: none"> • France: 2446-2483.5 • Japan: 2471-2497 • Spain: 2445-2475 <p>GFSK at a line rate of 1 Mbps</p> <ul style="list-style-type: none"> • Modulation Index 0.22 • Peak deviation 175 kHz • $BT = 0.5$
Frequency hopping	<ul style="list-style-type: none"> • 1600 hops/s • Four special hopping sequences reserved for connection setup • Periodicity of the sequence: 23 hours, 18 minutes
Transmit power (mW)	<ul style="list-style-type: none"> • Class 1: 1-100 • Class 2: 0.25-2.5 • Class 3: 1
Operating range (m)	<ul style="list-style-type: none"> • 0.1 to 10 • 100 with Class 1

GFSK is a form of digital frequency modulation using a Gaussian pulse with a modulation index that can be varied (if the index is equal to 0.5, GFSK becomes GMSK). In Bluetooth systems, a modulation index of 0.32 and a BT product of 0.5 are used. Other specifications of Bluetooth are given in Table 6.10 (Schin 2000). It is expected that Bluetooth-based products will be in the market shortly.

6.9 SUMMARY

The three multiple-access schemes, FDMA, TDMA, and CDMA, were discussed in this chapter. The two duplexing approaches, FDD and TDD, were briefly reviewed. Code division multiple access and the modulation schemes associated with CDMA, both direct-sequence and frequency hopping, were presented. The capacities of FDMA, TDMA, and CDMA were evaluated. The various modulation schemes and standards used in North America and Europe were compared. The G3 systems, cdma2000/1X and cdma2000/1X-T, were briefly discussed. An overview of Bluetooth systems was also provided.

- FDMA is a simple scheme. Each channel is allocated a frequency band. Different frequency bands are separated by a guardband. In the FDMA/FDD format, uplinks and downlinks use nonoverlapping frequency bands.
- FDMA is not compatible with variable-rate transmission. Power level balancing also is not easy. In addition, the transmitters and receivers must have a high Q value to ensure channel selectivity.
- TDMA systems are compatible with variable-rate transmission. However, synchronization issues are critical. TDMA schemes are also susceptible to fading.
- In CDMA each user is assigned a unique PN code. Each code consists of K chips, each with a duration of T_c and $KT_c = T$, the bit duration. Thus, CDMA uses a much larger bandwidth than TDMA or FDMA. All users share the same bandwidth all the time.

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- PN sequences are almost orthogonal to each other. This allows CDMA systems to have a "soft capacity limit" or no limit at all. However, as more and more users come into the system, the performance of the system goes down because of an increased level of interference.
- Since the chips are of extremely short duration, when multipath conditions exist the various delayed signals are resolvable, creating "multipath diversity." A RAKE receiver may be used to take advantage of the existence of multipath components.
- In DS-SS-CDMA systems, BPSK may be used as the fundamental modulation scheme.
- In FH-SS-CDMA, BPSK or MFSK is used.
- Fast frequency hopping occurs if more than one frequency hop is present during each transmitted symbol.
- Slow frequency hopping occurs if more than one symbol is transmitted between frequency hops.
- The capacity of CDMA is soft. It is possible for more users to come into the network. This may lead to a reduction in performance level.
- The capacity of CDMA is enhanced by the incorporation of voice activity detection schemes.
- The capacity of CDMA goes down when intercell interference is present.
- The future of wireless communications, cdma2000, is sketchy here.

PROBLEMS

1. Design a FDMA scheme that uses SSU-SC. If receiver noise is negligible, calculate the chip rate to maintain a bit error rate of 10^{-4} for a voice message is assumed to have a 4 kHz bandwidth. Draw the spectrum of the FDMA signal if each channel is allocated 5 kHz, including the guardband. The carrier frequency range is from 100 to 300 MHz. How many signals can be multiplexed?
2. Give a block diagram of a receiver to recover the signals generated in Problem 1. Show all the operations necessary to recover the message signal.
3. Consider data being generated at a rate of 1 Mbps. This data set is to be transmitted using DS-SS-CDMA with a processing gain of 8. Plot the data stream, a typical chip sequence, and the DS-SS-CDMA transmitted sequence.
4. Plot the spectra of the bit and the chip. Note the number of zero crossings in the spectra.
5. Using results of Problem 4, generate a BPSK-FSS signal.
6. In a DS-SS-CDMA cell, there are 64 equal power channels that share a common frequency band. The signal is transmitted using a BPSK format. The data rate is 3840 kbps. A coherent receiver is used for recovering the data. Assuming the receiver noise is negligible, calculate the chip rate to maintain a bit error rate of 10^{-4} .
7. In a DS-SS-CDMA system, the processing gain is 1000. The modulation format used is BPSK. If a bit error rate of 10^{-6} is required, how many users can share the system? Assume that the receiver noise is negligible.
8. A DS-SS-CDMA system is expected to have a processing gain of 30 dB. The expected data rate is 9600 bps. What should be the chip rate? (BPSK modulation will be employed, what is the theoretical maximum number of users using a multi-beam antenna?)
9. A shift register with 10 taps is used for generation of the PN sequence. If the chip duration is 0.1 μ s, calculate:
 - (a) The length of the PN sequence
 - (b) The bit duration
 - (c) The processing gain
10. A RAKE receiver will be used in the channel. If the delay of the multipath effects is 200 ns, what is the minimum delay necessary to successfully overcome the multipath effects? (Assume that the chip rate is 3840 kbps. A coherent receiver is used for recovering the data. Assuming the receiver noise is negligible, calculate the chip rate to maintain a bit error rate of 10^{-4} .)