The PN sequence generator is seeded with data received in messages sent from the base station. The seed is used to establish the voice and data privacy on the channel. The same seed is used in both directions.

5.7.9 Baseband Filtering

After PN modulation, the signal is filtered by a baseband filter. The filter should have the following parameters:

Passband ripple: 3 dB

Upper passband frequency: 590 kHz

Minimum stopband attentuation: 40 dB

Lower stopband frequency: 740 kHz

For the W-CDMA system, the following values hold (see table 5.9).

Table 5.9 Baseband Filter Parameters for W-CDMA System

System Bandwidth, MHz	Passband Ripple, dB	Upper Passband Frequency, MHz	Minimum Stopband Attenuation, dB	Lower Stopband Frequency, MHz
5.0	3	1.96	40	2.47
10.0	3	3.92	40	4.94
15.0	3	5.88	40	7.41

5.7.10 Synchronization of CDMA Signals

Time 0 for the CDMA system is January 6, 1980 at 00:00:00 UTC. This is the same as time 0 for the global positioning system (GPS); therefore, CDMA time is the same as GPS time. GPS time and UTC time differ by the number of leap seconds since January 6, 1980. Thus, GPS time and UTC time are synchronous but can differ by an integer number of seconds.

All BSs in a CDMA system are synchronized to the GPS. Each base station transmits a set of orthogonal codes that are synchronized to CDMA time and are time-shifted from the codes at other BSs in the system.

5.8 Summary

The International Standards Organization has developed the Open Systems Interconnect reference model for data communications that is used by most computer systems to design the communications protocols. In this chapter, we first reviewed the seven layers of the OSI model. We then limited discussion in this chapter to the physical layer (layer 1) of the CDMA and W-CDMA systems as defined in the ATIS and TIA standards. Both systems have the same goal: to efficiently use the available spectrum to provide digital cellular and PCS services. The CDMA system uses 64 orthogonal Walsh codes at a data rate of 1.2288 Mbps to code the digital signals for voice, data, and control. The W-CDMA system uses Walsh or Hadamard codes at higher bit rates (4.096, 8.192 and 12.288 Mbps) to accomplish the same result. The designers of the CDMA system do not attempt to recover pilot signal synchronization on the reverse channel (from mobile station to base station). They, therefore, use 64-ary modulation (with 1 of 64 Walsh symbols) on the reverse channel and use pseudorandom noise sequences to obtain the orthogonal modulation. The designers of the W-CDMA system believe that synchronization can be obtained on the reverse channel (at the higher data rates) and, therefore, use Walsh or Hadarmard codes in both directions. The two systems have other minor differences in the ordering between the various encoding steps but are otherwise similar.

Since the CDMA system can be placed anywhere in the cellular or PCS band, the standards define a set of preferred channels. A service provider can use any of the preferred channels. Because a critical component of CDMA systems is the ability to control the power of the mobile station almost instantaneously, we reviewed the power control methods used in both systems. We then concluded the chapter by defining the detailed modulation steps used in both systems and examined the differences between the two systems.

The CDMA system is defined by two standards: IS-95A for a dual-mode analog/digital system for cellular frequencies and J-STD-008 for a digital-only system at PCS frequencies. While there may be minor differences between IS-95A and J-STD-008 for the digital implementations, the goal of the two standards committees is that the digital part of both standards be identical except for frequency bands used. For the W-CDMA system, the standard is defined for PCS frequencies only, and one standard has two references numbers (IS-665 and J-STD-015). For more information on the specifics of the physical layer, consult the standards.

5.9 References

1. TIA IS-95A, "Mobile Station–Base Station Compatibility Standard for Dual Mode Spread Spectrum Cellular System."

2. Alliance for Telecommunications Industry Standards J-STD-008, "Personal Station–Base Station Compatibility Requirements for 1.8 to 2.0 GHz Code Division Multiple Access (CDMA) Personal Communications Systems."
3. TIA IS-665, "W-CDMA (Wideband Code Division Multiple Access) Air Interface Compatibility Standard for 1.85-1.99 GHz PCS Applications."
4. Alliance for Telecommunications Industry Standards J-STD-015, "W-CDMA (Wideband Code Division Multiple Access) Air Interface Compatibility Standard for 1.85-1.99 GHz PCS Applications."
5. "Radio System Characterization for the Proposed IS-95 based CDMA PCS Standard," Joint Technical Committee (on Air Interface Standards) of T1P1 and TR-46 contribution JTC(Air)/94.11.03-735, November 3, 1994.
6. Whipple, D. P., "The CDMA Standard," *Applied Microwave and Wireless*, 6(2), Spring 1994, pp. 24–37.
7. TIA PN-3570 (TSB-74), "Telecommunications Systems Bulletin: Support for 14.4 kbps Data Rate and PCS Interaction for Wideband Spread Spectrum Cellular Systems," October 1995.

CHAPTER 6

Network and Data Link Layers of CDMA

6.1 Introduction

In this chapter, we explore layers 2 and 3, the data link layer and the network layer used for CDMA, and the detailed messages that are sent over the CDMA air link. Some of the messages are sent only between the base station and the mobile station. Other messages are sent between the MS and other network elements. In chapter 7, we use these messages to show how call processing flows across the network. Information flows from the BS to the MS via the forward CDMA channel, on the pilot channel, the sync channel, the paging channel, and the forward traffic channel. Information flows from the MS to the BS on either the access channel or the reverse traffic channel. The BS/MS communications take place on the paging/access channel during call setup and on the forward/reverse traffic channel during a call.

All cellular and personal communications systems air interfaces (except GSM) used in North America share a common approach to the operation of a MS. In chapter 4, we discussed the high-level operation of the MS as it implements the common operational approach. In chapter 7, we discuss how the CDMA and W-CDMA systems use the operations described there to provide services; and in chapter 8, we discuss the voice coding systems used for CDMA.

Both CDMA and W-CDMA define control channels (sync, paging, and access channels) that are used for data communications between the MS and the PCS/cellular system and traffic channels that are used for user-to-user communications (voice or data).

The CDMA system combines the operation of the network and data link layers and treats them as one layer. The W-CDMA system uses higher-speed signaling and voice-encoding rates and takes an approach that is similar to ISDN and, thus, treats layers 2 and 3 as separate and distinct layers.

In this chapter, we discuss the detailed message framing for both systems and describe some of the typical messages that are sent in the system. There are many services supported in the CDMA system, and we encourage you to consult the applicable standards for a full treatment of the many messages.

When an MS is first powered up, it must find and decode data on a control channel before any further processing can be done. For the messages described in this chapter, we assume that the BS to MS channels are properly synchronized in the receivers of both sides and that the receivers are properly decoding data. The operations necessary for these events to happen are classified as engineering art and are usually proprietary to a given manufacturer of equipment. Some manufacturers provide integrated circuit chip sets to perform the proper data modulation and demodulation. The encoding and decoding of the messages described in this chapter are typically performed in the software (or firmware) of the BS and MS.

6.2 Forward CDMA Channel

Data can be transmitted from a BS to a MS over the sync channel, the paging channel, or the information stream on the forward traffic channel. Some of the data are specific to a particular channel. Other data (e.g., orders) can be sent on the paging channel or the traffic channel.

6.2.1 Sync Channel

The forward sync channel operates at a data rate of 1200 bps and transmits information that is specific to the BS and needed by the MS to access the system.

The Sync Channel message (fig. 6.1) has an 8-bit message length header, a message body of a minimum of 2 bits and a maximum of 1146 bits, and a cyclic redundancy check (CRC) code of 30 bits. If the sync channel messages are less than an integer multiple of 93 bits, they are padded with 0 bits at the end of the message. The message length includes the header, body, and CRC, but not the padding. The CRC is computed on the message length header and the message body using the following code:

$$g(x) = x^{30} + x^{29} + x^{21} + x^{20} + x^{15} + x^{13} + x^{12} + x^{11} + x^{8} + x^{7} + x^{6} + x^{2} + x + 1 \quad (6.1)$$

Message Length (in bytes)	Data	CRC	Padding = ...000...
8 bits	N_{MSG} = 2 - 1146 bits	30 bits	

Notes: N_{MSG} = Message length in bits (including length field and CRC)

Padding bits are not used for Unsynchronized Paging Channel Messages
Sync Channel Data Rate = 1200 bps
Paging Channel Data Rate = 4800 bps or 9600 bps

Figure 6.1 CDMA message framing on forward sync channel and paging channel.

After a message is formed, it is segmented into 31-bit groups and sent in a sync channel frame (fig. 6.2) consisting of a 1-bit start of message (SOM) field and 31 bits of the sync channel frame body. A value of 1 for SOM indicates that the frame is the start of a Sync Channel message. A value of 0 for SOM indicates that the frame is a continuation of a Sync Channel message or padding.

Three sync channel frames are combined to form a sync channel superframe (fig. 6.3) of length 80 ms (96 bits). The entire sync channel message is then sent in N superframes. The padding bits are used so that the start message always starts at one bit after the beginning of a superframe. The first bit of the superframe is SOM = 1.

The only message sent on the sync channel is the Sync Channel message that transmits information about the BS and the serving CDMA system. Some of the information being sent follows.

One set of data, the system identification (SID) and the network identification (NID), define the system being received and the network within the system. The values for SID and NID are defined by the Federal Communications Commission.

Other data define the offset of the PN sequence for the BS and the long code state for that BS.

The sync channel also sends information about the system time, leap seconds, offset from UTC, and the state of daylight savings time. These times can be used to provide an accurate clock in the MS and are also used to set the states of the various code generators in the MS.

Finally, the sync channel transmits information on the data rate used on the paging channel (4800 or 9600 bps).

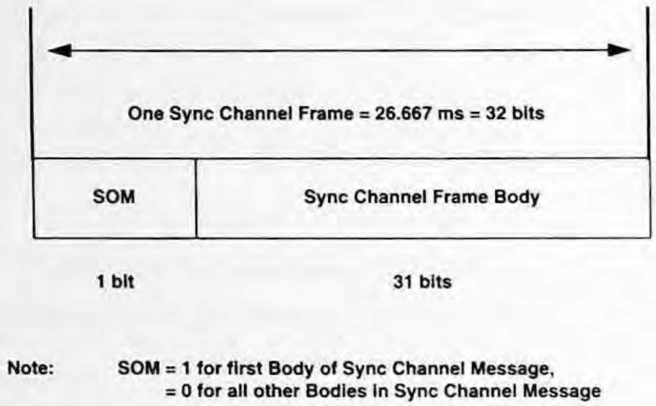


Figure 6.2 CDMA channel framing on forward sync channel.

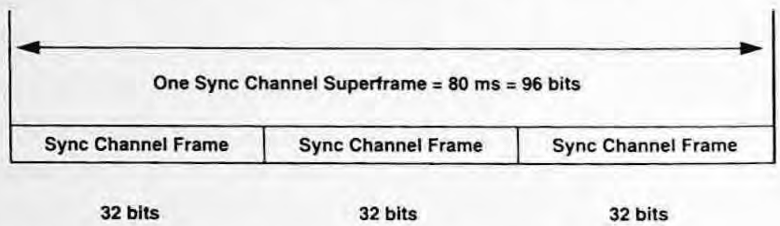


Figure 6.3 CDMA superframe structure on forward sync channel.

6.2.2 Paging Channel

The paging channel operates at a data rate of 4800 or 9600 bps and transmits overhead information, pages, and orders to an MS.

The Paging Channel message is similar in form to the Sync Channel message (fig. 6.1) and has an 8-bit message length header, a message body of a minimum of 2 bits and a maximum of 1146 bits, and a CRC code of 30 bits. The message length includes the header, body, and CRC, but not the padding. The CRC is computed on the message length header and the message body using the same code as the sync channel (equation 6.1).

Paging Channel messages can use synchronized capsules that end on a half-frame boundary or unsynchronized capsules that can end anywhere within a half-frame. If synchronized Paging Channel messages are less than an integer multiple of 47 bits for 4800-bps transmission (or 95 bits for 9600-bps transmission), they are padded with 0 bits at the end of the message. Unsynchronized messages do not have padding bits added to them.

After a message is formed, it is segmented into 47- or 95-bit chunks and sent in a sync channel half-frame (fig. 6.4) consisting of a 1-bit synchronized capsule indicator (SCI) field and 47 or 95 bits of the sync chan-

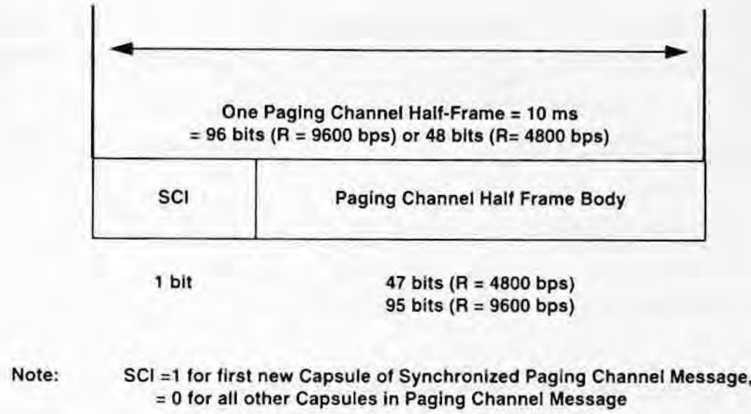


Figure 6.4 CDMA channel half-framing on forward paging channel.

nel frame body. A value of 1 for SCI indicates that the frame is the start of a Paging Channel message (either synchronized or unsynchronized). Messages can also start in the middle of a frame and immediately after the end of an unsynchronized message (with 0 padding bits). A value of 0 for SCI indicates that the frame is not the start of a message and can include a message (with or without padding), padding only, or the end of one message and the start of another.

Eight paging channel half-frames are combined to form a paging channel slot (fig. 6.5) of length 80 ms (384 bits at 4800 bps and 768 bits at 9600 bps). The entire Paging Channel message is then sent in N slots. The maximum number of slots that a message can use is 2048. The BS always starts a slot with a synchronized message capsule that starts at one bit after the beginning of a slot. The first bit in a slot is SCI = 1.

The paging channel sends many different types of messages; we mention a few and describe how they are used in chapter 7. We encourage you to consult the standards documents for a more detailed description of all of the messages. Some of the messages follow:

- **System Parameters Message:** This message is sent to all MSs in the area to specify the characteristics of the serving cellular/PCS system.
- **Access Parameters Message:** This message is sent to all MSs in the area to specify the characteristics of the messages sent on the access channel.
- **Order Message:** This message directs the MS to perform an operation and confirms a request from the MS. An example is an alerting message.

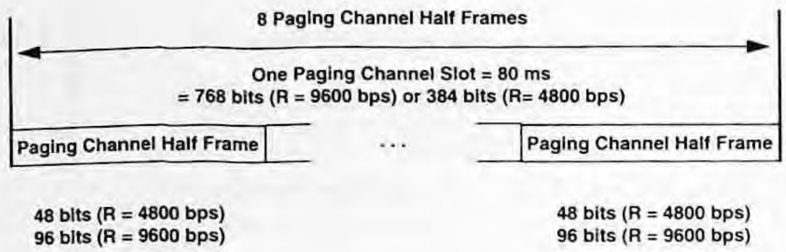


Figure 6.5 CDMA slot structure on the paging channel.

- **Channel Assignment Message:** This message informs the MS of the correct traffic channel to use for voice or data.
- **TMSI Assignment Message:** This message assigns a temporary mobile station identification (TMSI) to the MS. It is sent as part of the registration process described in chapter 7.

6.2.3 Traffic Channel

Channels not used for paging or sync can be used for traffic. The total number of traffic channels at a BS is 63 minus the number of paging and sync channels in operation at that BS.

Information on the traffic channels consists of primary traffic (voice or data), secondary traffic (data), and signaling in frames of length 20 ms.

When the data rate on the traffic channel is 9600 bps, each frame of 192 bits consists of 172 information bits, 12 frame quality bits, and 8 encoder tail bits (set to all 0s). At 4800 bps, there are 80 information bits, 8 frame-quality bits, and 8 tail bits for a total of 96 bits. At 2400 and 1200 bps, there are 40 and 16 information bits and 8 tail bits, for a total of 48 and 24 bits, respectively. The BS can select the data transmission rate on a frame-by-frame basis. The data rate of 9600 bps can support multiplexed traffic and signaling. Data rates of 1200, 2400, and 4800 bps can support only primary traffic information. The receiving MS determines the data rate being received by a combination of symbol error rates at each data rate and the frame quality data at the higher data rates.

The frame quality indicator is a CRC on the information bits in the frame. At 9600 bps the generator polynomial is

$$g(x) = x^{12} + x^{11} + x^{10} + x^{9} + x^{8} + x^{4} + x + 1 \tag{6.2}$$

At 4800 bps, the generator polynomial is

$$g(x) = x^{8} + x^{7} + x^{4} + x^{3} + x + 1 \tag{6.3}$$

At 9600 bps, the 172 information bits consist of 1 or 4 format bits and 171 or 168 traffic bits. A variety of different multiplexing options are supported. The entire 171 information bits can be used for primary traffic, or the 168 bits can be used for 80 primary traffic bits and 88 signaling traffic bits or 88 secondary traffic bits. Other options use 40 and 128 or 16 and 152 bits for primary and signaling/secondary traffic. Alternatively, the entire 168 bits can be used for signaling or secondary traffic.

When the forward traffic channel is used for signaling, the message is similar in form to the Paging Channel message (fig. 6.1) and has an 8-bit message length header, a message body of a minium of 16 bits and a maximum of 1160 bits, and a CRC code of 16 bits. Following the message are padding bits to make the message end on a frame boundary. The message length includes the header, body, and CRC, but not the padding. The CRC is computed on the message length header and the message body using the following code:

$$g(x) = x^{16} + x^{12} + x^{5} + 1 \tag{6.4}$$

When the forward traffic channel is used for signaling, some typical messages that can be sent follow:

- **Order Message:** This is similar to the order message sent on the paging channel.
- **Authentication Challenge Message:** When the BS suspects the validity of the MS, it can challenge the MS to prove its identity. We examine this in more detail in chapter 7.
- **Send Burst Dual-Tone Multifrequency (DTMF):** When the BS needs dialed digits, it can request them in this message. This message would be used for digits for a three-way call, for example.
- **Extended Handoff Direction Message:** This message is one of several handoff messages sent by the BS. See chapter 7 for more details on the handoff process.

6.3 Reverse CDMA Channel

The MS communicates with the BS over the access channel or the reverse traffic channel. The access channel is used to make originations, process orders, and respond to pages. After voice or data communications are established, all communications occur on the reverse traffic channel.

6.3.1 Access Channel

Whenever an MS registers with the network, processes an order, sends a data burst, makes an origination, responds to a page, or responds to an authentication challenge, it uses the (reverse) access channel.

The message on the reverse access channel consists of an access preamble of multiple frames of 96 zero bits with a length of 1 + PAM_SZ frames (fig. 6.6), followed by an access channel message capsule with length of 3 + MAX_CAP_SZ frames. The message capsule also consists of frames of length 96 bits. Since the data rate on the reverse access channel is 4800 bps, each frame has duration of 20 ms.

The entire access channel transmission therefore occurs in an access channel slot that has a length of

$$4 + \text{MAX\_CAP\_SZ} + \text{PAM\_SZ frames} \tag{6.5}$$

where the values of MAX_CAP_SZ and PAM_SZ are received on the paging channel.

An access channel slot nominally begins at a frame where

$$t \bmod(4 + \text{MAX\_CP\_SZ} + \text{PAM\_SZ}) = 0 \tag{6.6}$$

where t is the system time in frames.

The actual start of the transmission on the access channel is randomized to minimize collisions between multiple MSs accessing the channel at the same time.

All access channels corresponding to a paging channel have the same slot length. Different BSs may have different slot lengths.

The Access Channel message (fig. 6.7) is similar in form to the Sync Channel message and has an 8-bit message length header, a message body of a minimum of 2 bits and a maximum of 842 bits, and a CRC code of 30 bits. Following the message are padding bits to make the message end on a frame boundary. The message length includes the header, body, and CRC, but not the padding. The CRC is computed on the message length header and the message body using the same code as the sync channel (equation 6.1).

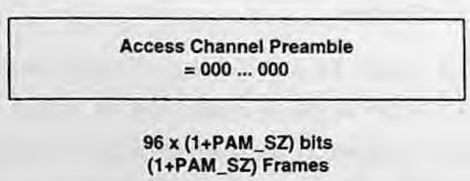


Figure 6.6 CDMA access channel preamble.

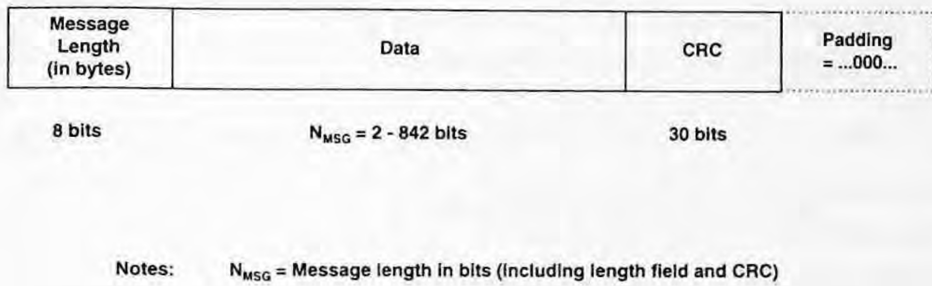


Figure 6.7 CDMA message framing on access channel.

Each access channel frame contains either preamble bits (all zeros) or message bits. Frames containing message bits (fig. 6.8) have 88 message bits and 8 encoder tail bits (set to all zeros). Multiple frames are combined with an access channel preamble to form an access channel slot (fig. 6.9).

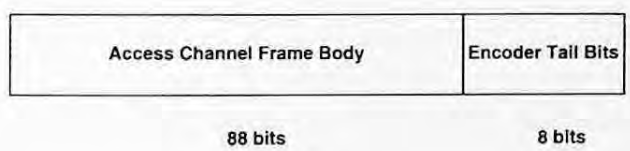


Figure 6.8 CDMA access channel framing.

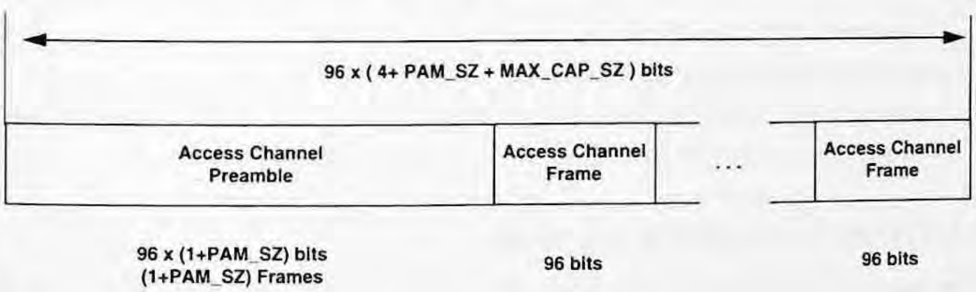


Figure 6.9 CDMA access channel slot.

6.3.2 Traffic Channel

Information on the reverse traffic channels consists of primary traffic (voice or data), secondary traffic (data), and signaling using frames of length 20 ms.

The message format is identical to the forward traffic channel. When the reverse traffic channel is used for signaling, the message (fig. 6.10) has an 8-bit message length header, a message body of a minimum

Message Length (In bytes)	Data	CRC	Padding = ...000...
8 bits	N_{MSG} = 16 - 2016 bits	16 bits	

Notes: N_{MSG} = Message length in bits (including length field and CRC)

Figure 6.10 CDMA message framing on reverse traffic channel.

of 16 bits and a maximum of 2016 bits, and a CRC code of 16 bits. Padding bits follow the message to make the message end on a frame boundary. The message length includes the header, body, and CRC, but not the padding. The CRC is computed on the message length header and the message body using the code described in equation (6.4).

When the reverse traffic channel is used for signaling, some of the following example messages can be sent:

- **Order:** This message is either a response to a BS request or a request for service from the MS.
- **Authentication Challenge Response:** This message is sent in response to the challenge by the BS.
- **Flash with Information:** When the user requires special services from the BS, a flash message is sent. This messages is similar to depressing the switch-hook on a wireline phone. The message may or may not contain additional information.
- **Handoff Completion:** When the MS completes the handoff process, it sends this message.

6.4 Forward W-CDMA Channel

The operation of the forward channel (BS to MS) in W-CDMA is similar to that of CDMA. However, the W-CDMA system separates the operation of layers 2 and 3 of the OSI protocol stack. Messages on the forward and reverse channel are formed at layer 3 and passed to layer 2. At layer 2, the messages are formed into a frame and sent to the physical layer (chapter 5).

The goal of the W-CDMA system is to model the operation of ISDN. Although the detailed ISDN message set is not used, the signaling data rates and voice-encoding rates are compatible with ISDN.

Data can be transmitted from a BS to an MS over the sync channel, paging channel, or information stream on the forward traffic channel.

Some of the data are specific to a particular channel. Other data (e.g., orders) can be sent on the paging channel or the traffic channel.

6.4.1 Layer-to-Layer Communications

The W-CDMA system defines a set of primitives for layer-to-layer communications to more closely implement the OSI reference model. Primitives are defined for layer 3-to-layer 2 communications and for layer 2-to-layer 1 communications. Primitives are also defined for management functions between layers. We examine the layer-to-layer communications here and refer you to the standard for the management functions.

In the CCITT layer control, four basic functions are defined for any primitive function (see fig. 6.11):

- **Request:** The higher layer makes a request to the lower layer. This request is passed to the receiving side.
- **Indication:** The lower layer at the receiving side passes an indication primitive to its next higher layer.
- **Response:** The higher layer on the receiving side performs an action (or requests an action from its higher layer) and sends a response message when the action is complete.
- **Confirm:** The lower layer passes the response to the transmitting side where the lower layer passes a confirmation (of the request) to its upper layer.

Not all primitives implement all four of the basic functions. Some implement only Request and Indication, for example.

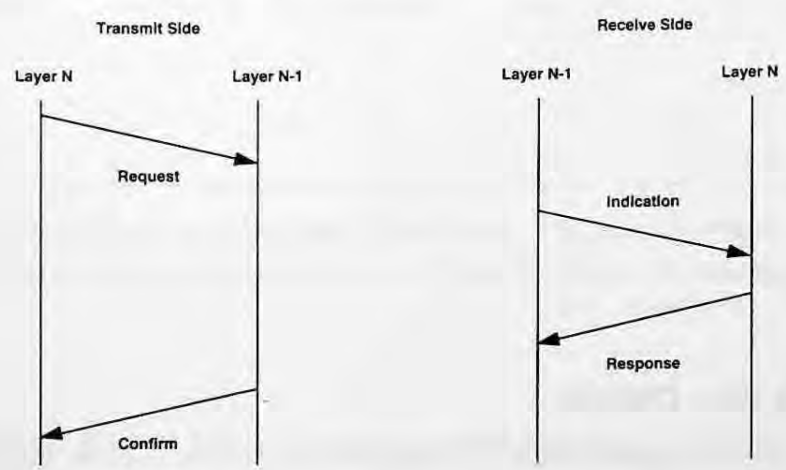


Figure 6.11 Interlayer primitives.

For layer 3-to-layer 2 communications, five primitives are defined (see table 6.1). Data link establish (DL-Establish) and release (DL-Release) are used to start and stop multiframe connection-oriented messages. Data link data (DL-Data) and Data link unit data (DL-Unit-Data) are used to transmit and receive acknowledged and unacknowledged messages, respectively. Data link setup (DL-Setup) is used to transmit and receive connectionless messages.

Table 6.1 Layer 3-to-Layer 2 Primitives

Primitive	Primitive Function			
	Request	Indication	Response	Confirm
DL-Establish	X	X	—	X
DL-Release	X	X	—	X
DL-Data	X	X	—	—
DL-Unit Data	X	X	—	—
DL-Setup	X	X	X	X

For layer 2-to-layer 1 communications, three primitives are defined (see table 6.2). Physical layer active (PH-Active) and physical layer deactivate (PH-Deactivate) are used to establish and release a physical layer connection. Physical data (PH-Data) is used to pass messages between layer 2 and layer 1.

Table 6.2 Layer 3-to-Layer 2 Primitives

Primitive	Primitive Function			
	Request	Indication	Response	Confirm
PH-Data	X	X	—	—
PH-Active	X	X	—	—
PH-Deactive	—	X	—	—

The layer 2-to-layer 3 primitives and the layer 2-to-layer 1 primitives are used in both the forward and reverse direction on the W-CDMA channel.

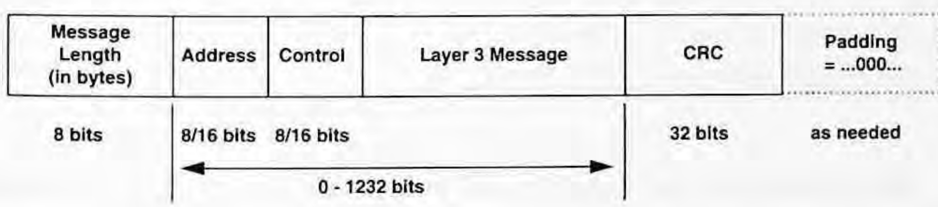
6.4.2 Sync Channel

The forward sync channel operates at a data rate of 16 kbps and transmits information that is specific to the BS and needed by the MS to access the system.

The Sync Channel message (fig. 6.12) has an 8-bit message length header, an 8- or 16-bit address field, an 8- or 16-bit control field, a layer 3 message body of a minimum of 0 bits and a maximum of 1232 bits, and a CRC code of 32 bits. If the Sync Channel messages are less than an integer multiple of 319 bits, they are padded with 0 bits at the end of the message. The message length includes the length field, the address field, the control field, the body, and CRC, but not the padding. The CRC is computed on the message length header, the address, the control, and the message body using the following code:

$$g(x) = x^{32} + x^{26} + x^{23} + x^{22} + x^{16} + x^{12} + x^{11} + x^{10} + x^{8} + x^{7} + x^{5} + x^{4} + x^{2} + x + 1 \quad (6.7)$$

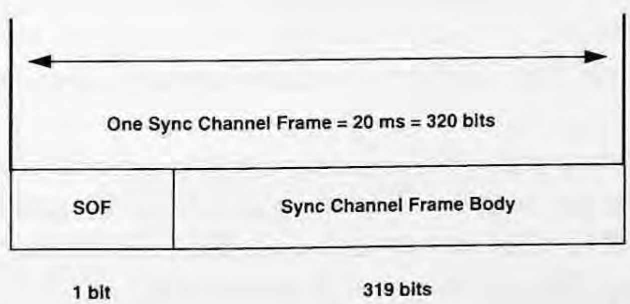
After a message is formed, it is segmented into 319-bits groups and sent in a sync channel frame (fig. 6.13) consisting of a 1-bit start of frame



Notes: Message length (in bytes) includes: length field, address, control, layer 3 message and CRC)

Padding bits are not used for Unsynchronized Paging Channel Messages
Sync Channel Data Rate = 16 kbps
Paging Channel Data Rate = 16 kbps

Figure 6.12 W-CDMA message framing on forward sync channel and paging channel.



Note: SOF = 1 for first Body of Sync Channel Message,
= 0 for all other Bodies in Sync Channel Message

Figure 6.13 W-CDMA channel framing on forward sync channel.

(SOF) field and 319 bits of the sync channel frame body. A value of 1 for SOF indicates that the frame is the start of a sync channel message. A value of 0 for SOF indicates that the frame is a continuation of a sync channel message or padding.

Four sync channel frames are combined to form a sync channel superframe (fig. 6.14) of length 80 ms (1280 bits). The entire sync channel message is then sent in N superframes. The padding bits are used so that the start message always starts at one bit after the beginning of a superframe. The first bit of the superframe is SOF = 1.

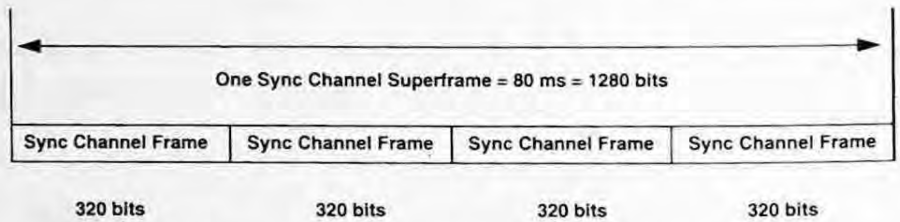


Figure 6.14 W-CDMA superframe structure on forward sync channel.

Two messages are sent on the sync channel: a mobility management (MM) identification message and a System Sync message. These two messages provide information similar to the single Sync Channel message of the CDMA system.

The System Sync message provides information about the number of 20-ms slots used for messages on the paging channel and the frequency allocation of the W-CDMA system (i.e., the bandwidth and channels used). The System Sync message defines the Walsh or Hadamard set used for the primary paging channel, the index into that set for the BS, and the pilot PN offset used by the BS. The message also informs the MS about the system date and time and the protocol revision supported by the BS.

The MM message provides the random number (RAND) used for authentication (see chapter 7), the system ID (SID), and the registration zone (REG_ZONE). The use of SID and REG_ZONE, for the purposes of paging roaming MSs, is discussed in chapters 4 and 7.

6.4.3 Paging Channel

For W-CDMA, the paging channel operates at a fixed data rate of 16 kbps and transmits overhead information, pages, and orders to a MS.

The Paging Channel message is identical in form to the sync channel message (fig. 6.12).

The Paging Channel messages are sent in paging channel slots (fig. 6.15) of length 80 ms (1280 bits). The first bit of the slot is the start of slot (SOS) bit and the other 1279 bits contain the message. A group of 32 contiguous paging channel slots form a paging channel slot cycle of length 2.56 seconds (see fig. 6.16). The value of 1 for SOS indicates the start of a slot cycle. The other 31 slots in the cycle have value of 0 for SOS.

Paging Channel messages are sent in one or more slots. The message is padded to end on a slot boundary. Multiple short messages can be contained within a single slot or a long message can extend across multiple slots.

The types of messages sent on the forward paging channel are similar to those sent in the CDMA system. There are some exceptions; for example, the broadcast message is used to send information about the system and provides information similar to the system parameters message in the CDMA system. Chapter 7 describes the messages for the specific applications described there.

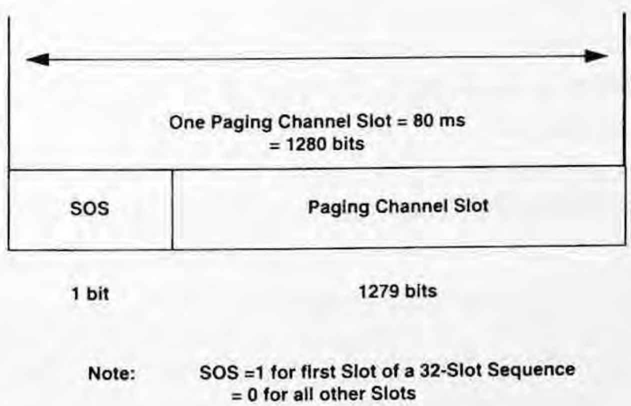


Figure 6.15 W-CDMA paging channel slots.

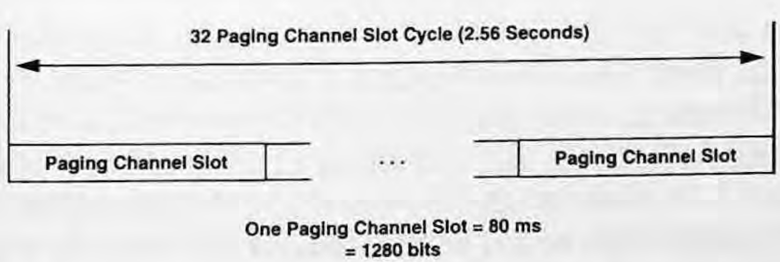


Figure 6.16 W-CDMA paging channel slot cycle.

6.4.4 Traffic Channel

The forward traffic channel operates at an effective data rate of 64 kbps. The signaling traffic, when sent, is at either 2 or 4 kbps. In an addition, there is a 2-kbps power control channel. The W-CDMA system supports voice or data traffic at rates of 16, 32, and 64 kbps. Chapter 7 discusses the voice coding for CDMA and W-CDMA. The entire traffic data stream of voice or data, signaling, and power control is multiplexed into a 64-kbps channel. The multiplexing is performed by a combination of the interleaver, symbol repetition, and multiplexer described in chapter 5 and figure 5.9.

When signaling is needed on the forward traffic channel, the messages that are sent are similar to those on the CDMA forward traffic channel. Specific examples of messages used are discussed in chapter 7.

6.5 Reverse W-CDMA Channel

The reverse W-CDMA functions are similar to those on the reverse CDMA channel but operate at a higher data rate. The MS communicates with the BS over the access channel or the reverse traffic channel. The access channel is used to make originations, process orders, and respond to pages. After voice or data communications are established, all communications occur on the reverse traffic channel.

The reverse channel implements the same layer 2 and layer 3 approach that is implemented on the forward channel.

6.5.1 Layer-to-Layer Communications

The same primitives described in section 6.4.1 for the forward channel are used for the reverse channel.

6.5.2 Access Channel

Whenever an MS registers with the network, processes an order, sends a data burst, makes an origination, responds to a page, or responds to an authentication challenge, it uses the (reverse) access channel.

The message on the reverse access channel consists of an access preamble of 1–3 frames of 320 zero bits (fig. 6.17), followed by an Access Channel message with length of 3 to 1 frames (fig. 6.18), for a total message length (fig. 6.19) of 4 frames (1280 bits). Since the data rate on the reverse access channel is 64 kbps, each frame has duration of 20 ms.

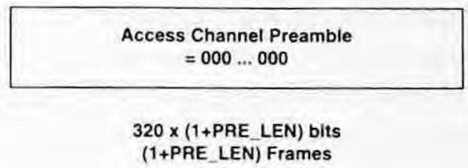


Figure 6.17 W-CDMA access channel preamble.

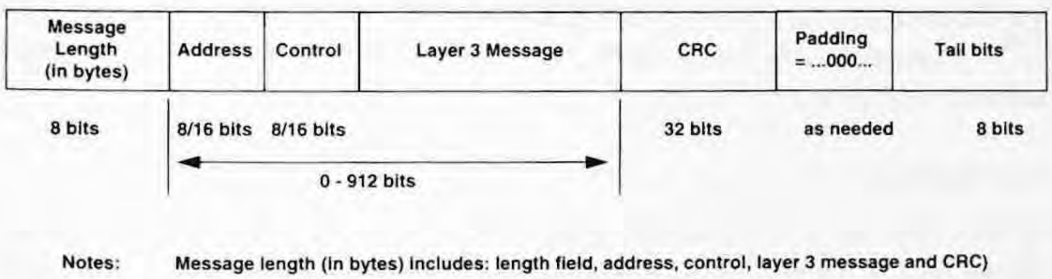


Figure 6.18 W-CDMA message framing on access channel.

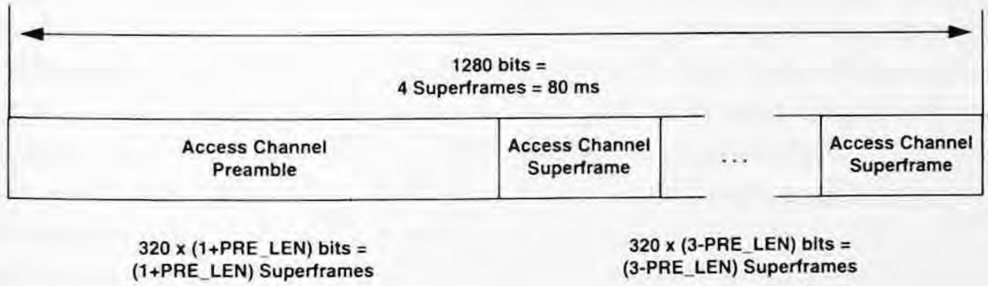


Figure 6.19 CDMA access channel slot.

The actual start of the transmission on the access channel is randomized to minimize collisions between multiple MSs accessing the channel at the same time.

The Access Channel message (fig. 6.18) is similar in form to the Sync Channel message and has an 8-bit message length header, an address field of 8 or 16 bits, a control field of 8 or 165 bits, a message body of a minimum of 0 bits and a maximum of 912 bits, and a CRC code of 32 bits. Following the message are padding bits and 8 encoder tail bits. The padding bits are added to make the message end on a frame boundary. The message length includes the header, address, control, body, and CRC, but not the padding and tail bits. The CRC is computed on the message length header and the message body using the same code as the sync channel equation (6.7).

Each access channel transmission consists of 1280 bits made up of four 320 bit frames (fig. 6.19). Each frame contains either preamble bits (all zeros) or message bits. Frames containing message bits (fig. 6.18) have 8 encoder tail bits (set to all 0s) at the end of the frame.

6.5.3 Traffic Channel

The format of the reverse traffic channel is identical to the forward traffic channel, except that the pilot signal is sent at reduced power.

The messages are similar to those on the reverse CDMA channel and are discussed in chapter 7.

6.6 Summary

In this chapter, we examined the structure of the layer 2 and layer 3 messages on the forward (BS to MS) and reverse (MS to BS) channels for both the CDMA and W-CDMA systems. The CDMA system combines layers 2 and 3, whereas the W-CDMA system segments the functionality of layers 2 and 3. The forward channels are the sync channel, the paging channel, and the traffic channel. On the reverse link, the channels are the access channel and the traffic channel. The sync channel establishes the synchronization of the received data at the MS receiver. The paging channel is used to page the MS and send it orders. The reverse access channel enables the MS to respond to orders and originate calls. The traffic channel is used for digital voice or data. Both systems permit control information to be multiplexed into the traffic channel stream to send and receive data between the MS and the BS.

6.7 References

1. TIA IS-95A, "Mobile Station–Base Station Compatibility Standard for Dual Mode Spread Spectrum Cellular System."
2. Alliance for Telecommunications Industry Standards J-STD-008, "Personal Station–Base Station Compatibility Requirements for 1.8 to 2.0 GHz Code Division Multiple Access (CDMA) Personal Communications Systems."
3. TIA IS-665, "W-CDMA (Wideband Code Division Multiple Access) Air Interface Compatibility Standard for 1.85-1.99 GHz PCS Applications."
4. Alliance for Telecommunications Industry Standards J-STD-015, "W-CDMA (Wideband Code Division Multiple Access) Air Interface Compatibility Standard for 1.85-1.99 GHz PCS Applications."

CHAPTER 7

Signaling Applications in the CDMA System

7.1 Introduction

This chapter discusses the signaling applications in the CDMA system. The data application is discussed in chapter 10. As we described in chapter 5, the actual application is above the layer 7 software. All the U.S.-based cellular and PCS air interfaces share a common heritage from two sources. The original analog cellular system was defined in 1979 by the EIA. All the digital air interfaces, including CDMA and wideband CDMA, inherit their characteristics from the analog protocol [10]. In addition, they all have services based on the IS-41 standard for intersystem communications. IS-41 and IS-104 (for supplementary services) define the functionality for a mobile application part (MAP) for U.S.-based PCSs. They also use mobility management applications part (MMAP) for signaling from the base station to the mobile switching center. This signaling is based on ISDN.

In chapter 4, we discussed the TR-45/TR-46 model and the basic and supplementary services supported in a cellular/personal communication system. In this chapter, we will discuss the end-to-end call flow for some typical basic and supplementary services. These end-to-end call flows are synthesized from examination of the various standards and do not appear in any one document within the standards.

7.2 End-to-End Operation of a Wireless System

This section describes the operation of a wireless system. We trace call flows from a mobile station to a base station to the MSC to other network elements. The flows are based on the TR-45/TR-46 reference model and an A-interface based on the Integrated Services Digital Network. The ISDN model assumes that there are ISDN terminals associated with the switch, one for each directory number on the switch. Since a cellular or personal communications system allows mobile stations to be associated with any base station, there is not a one-for-one correspondence between MSs and ISDN terminals. Thus, with the ISDN model, each MS registered at a base station is assigned a temporary directory number (also called virtual terminal number or interface directory number) that the radio system and PCS switch use to refer to the MS in the ISDN signaling messages, while that MS is registered at the base station. The ISDN A-interface defines a PCS application protocol (PCSAP) that uses ISDN signaling. With basic ISDN and PCSAP, the MSC can support terminal mobility. The radio system and switch can interact in either of two methods. In method one, the radio system is equivalent to a private branch exchange (PBX) and the switch is an ISDN switch that does minimal call control. In method two, the radio system is a virtual ISDN terminal with all call control in the ISDN switch. We have showed call flows for method two and leave the construction of call flows for method one as an exercise for you. Either method will work, although some systems designers will support one method over the other.

In chapter 3, we discussed the basic call-processing functions in the CDMA and wideband CDMA phones, and in chapter 6, we discussed some of the messages that are sent over the radio link. We will now use those functions and messages to describe the end-to-end call flows for several voice-based telecommunications services. In chapter 10, we will discuss wireless data services. We assume that you have studied the TR-45/TR-46 reference model described in chapter 4 and will refer to the various elements in that reference model as we describe the call flows.

7.2.1 Basic Services

Before a mobile station can originate or receive a call, it will register with the wireless system. An exception is made for emergency (911) calls. During the registration process, the MS is given a temporary mobile station identity (TMSI)[1] that is used for all subsequent call processing. We will first discuss the registration process and then call flows

1. Most cellular phones and some early PCS phones do not support TMSI and therefore will use a Mobile Identification Number (MIN) to identify themselves to the network.

for other services. Except for emergency calls, the mobile station must be registered on the PCS to receive services. The following call flows are based on an ISDN A-interface between the MSC and the BS. The call flows presented here are representative of the MS, BS, MSC, and PSTN network interactions. Many equipment vendors support proprietary interfaces between the MSC and BS. The call flows for their systems may be different within the network.

7.2.1.1 Registration *Registration* is the means by which a mobile station informs a service provider of its presence in the system and its desire to receive service from that system. The MS may initiate registration for several different reasons.

A mobile station registering on an access channel may perform any of the following registration types:

- **Distance-Based Registration** is done when the distance between the current base station and the base station where the MS last registered exceeds a threshold.
- **Ordered Registration** is done when the system sets parameters on the forward paging channel to indicate that all or some of the MSs must register. The registration can be directed to a specific MS or a class of MSs.
- **Parameter Change Registration** is done when specific operating parameters in the MS are changed.
- **Power-Down Registration** is done when the mobile station is switched off. This allows the network to deregister a mobile station immediately upon its power-down.
- **Power-Up Registration** is done when power is applied to the mobile station and is used to notify the network that the MS is now active and ready to place or receive calls.
- **Timer-Based Registration** is done when a timer expires in the mobile station. This procedure allows the data base in the network to be cleared if a registered MS does not reregister after a fixed time interval. The time interval can be varied by setting parameters on the control channel.
- **Zone-Based Registration** is done whenever an MS enters a new area of the same system. A service area may be segmented into smaller regions, location areas, or zones, which are a group of one or more cells. The MS identifies the current location area via parameters transmitted on the forward paging channel. Location-based

registration reduces the paging load on a system by allowing the network to page only in the location area(s) where a mobile station is registered.

Two other forms of registration occur when the mobile station takes certain actions:

- **Implicit Registration** occurs when a mobile station successfully communicates with the base station for a page response or an origination.
- **Traffic Channel Registration** occurs when the mobile station is assigned a traffic channel. The base station can notify the MS that it is registered.

In chapter 3, we discussed the types of registration that CDMA and W-CDMA system implement.

The following steps are the call flows for the registration of all mobile stations listening to a paging channel (see fig. 7.1):

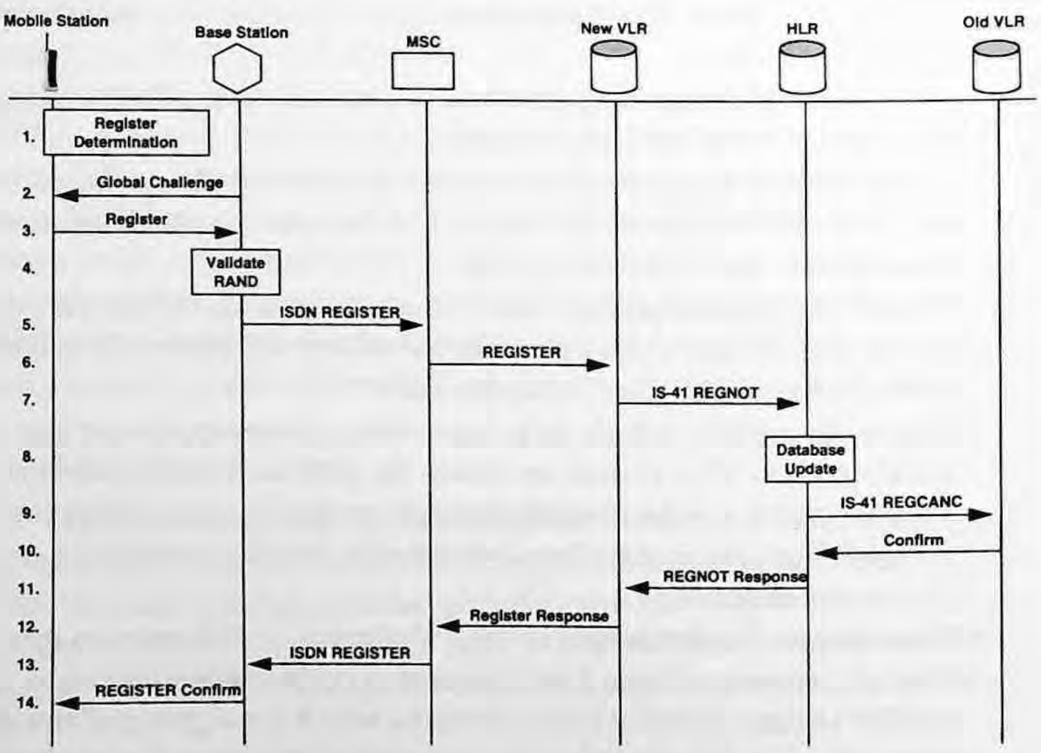


Figure 7.1 Mobile station registration.

1. The MS determines that it must register with the system.
2. The MS listens on the paging channel for the global challenge, RAND.
3. The MS sends a message to the base station with its international mobile station identification (IMSI), RAND, response to the challenge (AUTHR), and other parameters, as needed, in the MS registration request.
4. The base station validates RAND.
5. The base station sends an ISDN REGISTER message to the MSC.
6. The MSC receives the REGISTER message and sends a message to the serving VLR.
7. If the MS is not currently registered to the serving VLR, the VLR sends a REGistration NOTification (REGNOT) message to the user's HLR containing the IMSI, and other data as needed.
8. The MS's HLR receives the REGNOT message and updates its data base accordingly (stores the location of the VLR that sent the REGNOT message).
9. The MS's HLR sends an IS-41 REGistration CANCel (REGCANC) message to the old VLR where the MS was previously registered so that the old VLR can cancel the MS's previous registration.
10. The old VLR returns a confirmation message that includes the current value of call history count (CHCNT).
11. The user's HLR then returns a REGNOT response message to the (new) visited VLR and passes along information that the VLR needs (e.g., user's profile, interexchange carrier ID, shared secret key for authentication, and current value of CHCNT). If the registration is a failure (due to invalid IMSI, service not permitted, nonpayment of bill, etc.), then the REGNOT response message will include a failure indication.
12. Upon receiving the REGNOT response message from the user's HLR, the VLR assigns a temporary mobile station identification (TMSI) and then sends a registration notification response message to the MSC.
13. The MSC receives the message, retrieves the data, and sends an ISDN REGISTER message to the base station.
14. The base station receives the REGISTER message and forwards it to the MS to confirm registration.

Some CDMA systems will support the sending of the old TMSI when an MS registers in a new system. When an MS sends its old TMSI, the call process flow is similar except that the new VLR communicates with the old VLR to obtain the IMSI before an HLR query can be done.

7.2.1.2 Call Origination *Call origination* is the service wherein the MS user calls another telephone on the world-wide telephone network. It is a cooperative effort among the MSC, the VLR, and the base station.

The detailed call flow steps follow (see fig. 7.2 for the call flow diagram):

1. The MS processes an Origination Request from the user and sends it to the base station.
2. The base station sends a PCSAP Qualification Request to the VLR.
3. The VLR returns a Qualification Request Response to the base station.

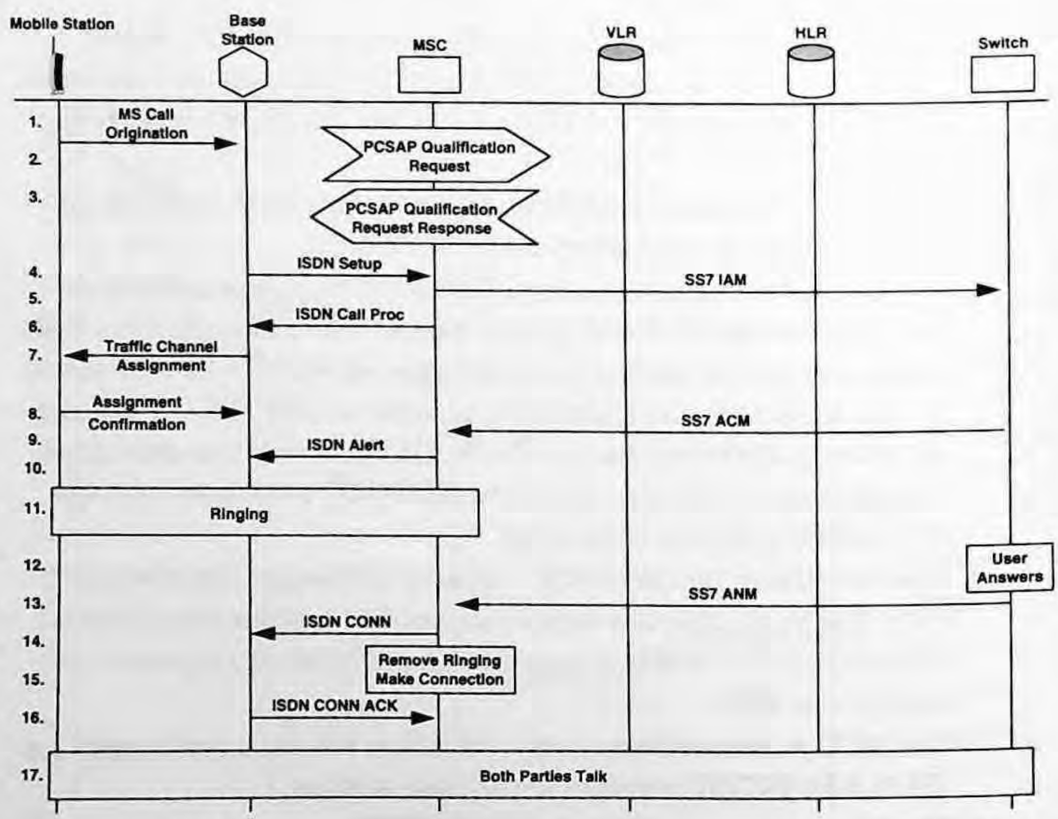


Figure 7.2 Mobile station call origination.

4. The base station then processes an ISDN Setup message and sends it to the MSC.
5. The MSC sends an SS7 (ISDN User Part) Initial Address message (IAM) to the terminating switch (wireline or wireless).
6. At the same time, the MSC returns an ISDN Call Proceeding message to the base station.
7. The base station assigns a traffic channel to the MS.
8. The MS tunes to the traffic channel and confirms the traffic channel assignment.
9. The terminating switch checks the status of the called telephone and returns an SS7 Address Complete message (ACM) to the MSC.
10. The MSC returns an ISDN alert message to the base station.
11. The MSC provides audible ringing to the user.
12. The terminating user answers.
13. The terminating switch sends an SS7 ANswer message (ANM) to the MSC.
14. The MSC sends and ISDN CONNect message to the base station.
15. The MSC removes audible ringing and makes the network connection.
16. The base station returns an ISDN CONNect ACKnowlege message.
17. The two parties establish their communications.

7.2.1.3 Call Termination *Call termination* is the service wherein an MS user receives a call from other telephones in the world-wide telephone network. The following discussion is for calls terminating to a MS registered at its home MSC. Calls terminating to roaming MSs will be discussed in section 7.2.1.5. Call termination is a cooperative effort among the MSC, the VLR, and the base station.

The detailed call flow steps follow (see fig. 7.3 for the call flow diagram):

1. A user in the world-wide phone network (wired or wireless) dials the directory number (DN) of the MS.
2. The originating switch sends an SS7 IAM to the MSC.
3. The MSC queries the VLR for the list of radio systems (one or more) where the MS will be paged and for the TMSI of the MS.
4. The VLR returns with the TMSI and a list of base stations.

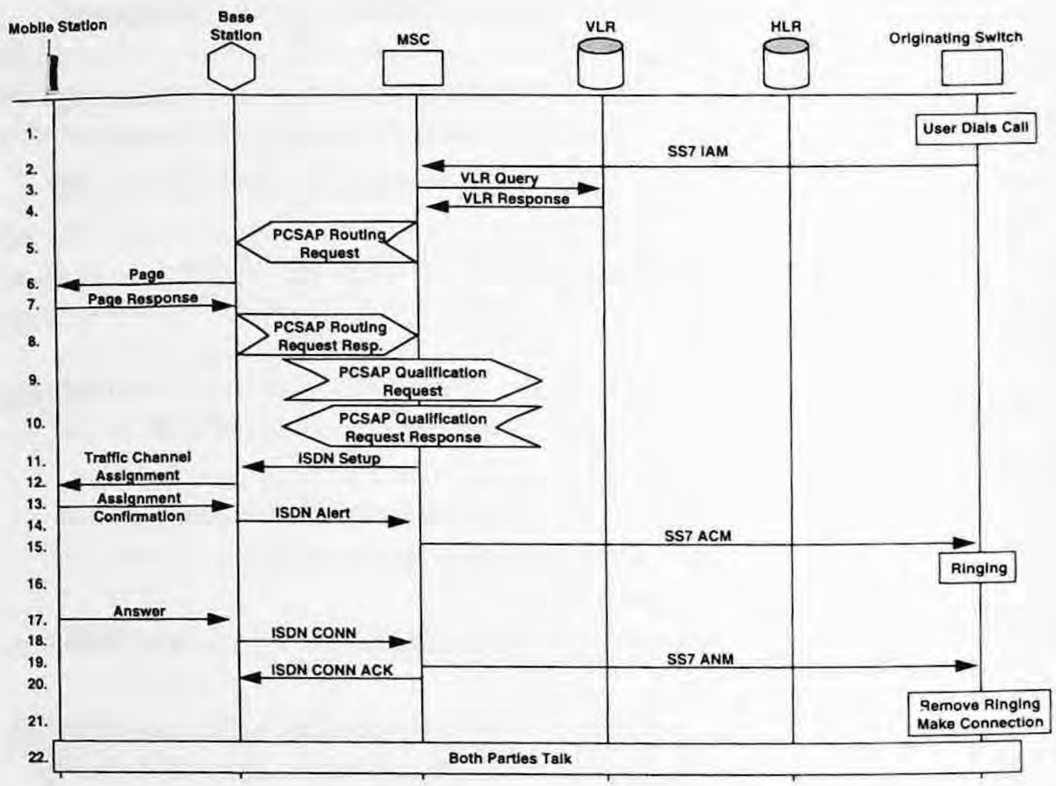


Figure 7.3 Call termination to a mobile station.

5. The MSC sends an PCSAP Routing Request message to all base stations on the list.
6. Each base station broadcasts a Page message on appropriate paging channels.
7. The MS responds to the page with a Page Response message at one base station.
8. The base station sends a PCSAP Routing Request Response to the MSC.
9. The base station sends a PCSAP Qualification Directive message to the VLR.
10. The VLR responds with a PCSAP Qualification Request Response.
11. The MSC sends an ISDN Setup message to the base station.
12. The base station sends a traffic channel assignment to the MS.
13. The MS tunes to the traffic channel and sends a Traffic Channel Assignment Confirmation message.
14. The base station sends an ISDN Alert message to the MSC.

15. The MSC sends an SS7 ACM to the originating switch.
16. The originating switch applies audible ringing to the network.
17. The user answers, and the MS sends a response message to the base station.
18. The base station sends an ISDN CONNect message to the MSC.
19. The MSC sends an SS7 ANswer Message (ANM) to the originating switch.
20. The MSC sends and ISDN CONNect ACKnowlege message to the base station.
21. Audible ringing is removed.
22. The two parties establish their communications.

7.2.1.4 Call Clearing When either party in a conversation wishes to end a call, then the call clearing function is invoked. The exact call flows depend on which side ends the call first. It is a cooperative effort among the MSC, the VLR, and the base station.

The detailed call flow steps, for an MS-initiated call clearing follow (see fig. 7.4 for the call flow diagram):

1. The MS user hangs up.

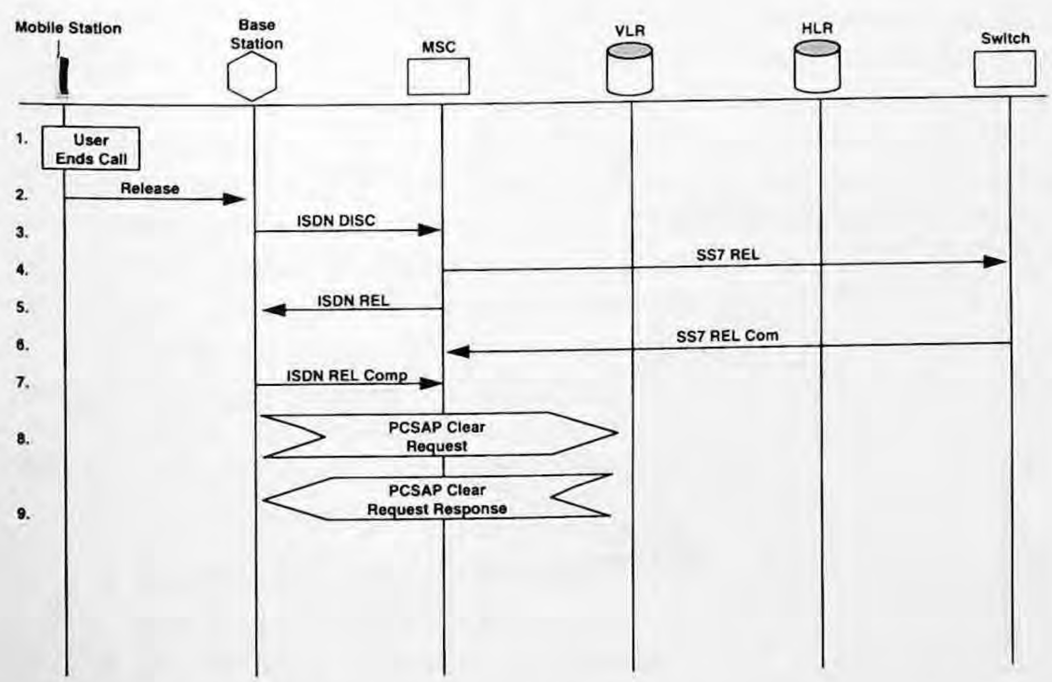


Figure 7.4 Call clearing—mobile station initiated.

2. The MS sends a Release message to the base station.
3. The base station sends an ISDN DISConnect message to the MSC.
4. The MSC sends an SS7 RELease message to the other switch.
5. The MSC sends an ISDN RELease message to the base station.
6. The other switch sends an SS7 RELease Complete message to the MSC.
7. The base station sends an ISDN RELease Complete message to the MSC.
8. The base station sends a PCSAP Clear Request message to the VLR.
9. The VLR closes the call records and sends a PCSAP Clear Request Response message to the base station.

The detailed call flow steps, for a far end–initiated call clearing follow (see fig. 7.5 for the call flow diagram):

1. The far end user hangs up.
2. The far end switch sends an SS7 RELease message to the MSC.
3. The MSC sends an ISDN DISConnect message to the base station.
4. The base station sends a Release message to the MS.

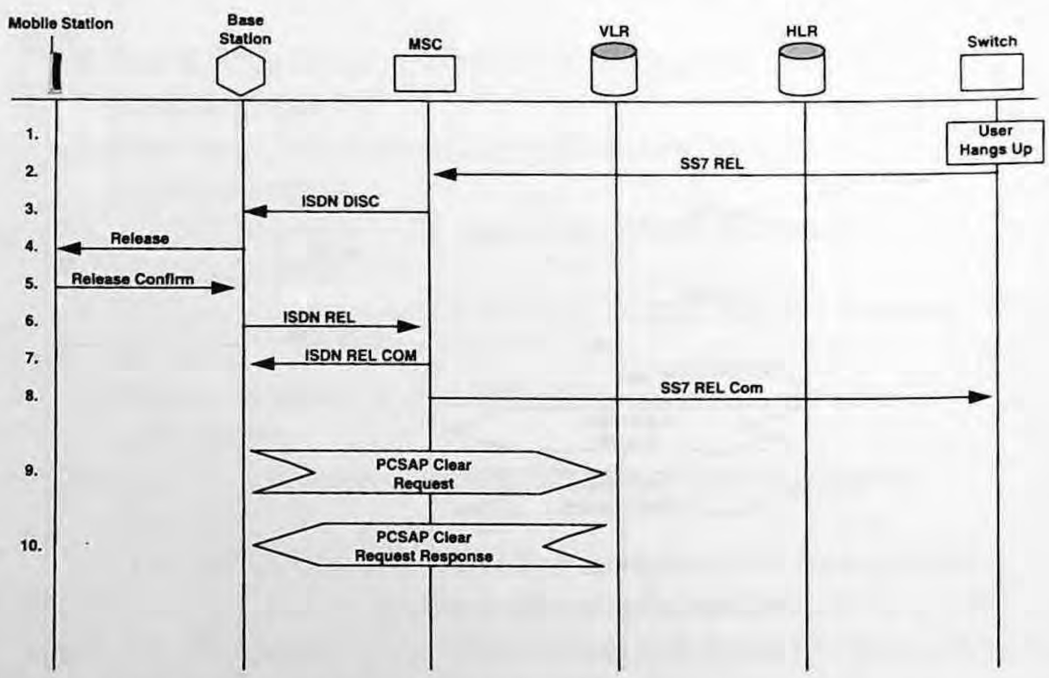


Figure 7.5 Call clearing—far end initiated.

5. The MS confirms the message and disconnects from the traffic channel.
6. The base station sends an ISDN RELease message to the MSC.
7. The MSC sends an ISDN RELease COMplete message to the base station.
8. The MSC sends an SS7 RELease COMplete message to the other switch.
9. The base station sends a PCSAP Clear Request message to the VLR.
10. The VLR closes the call records and sends a PCSAP Clear Request Response message to the base station.

7.2.1.5 Roaming *Roaming* is the ability to deliver services to mobile stations outside of their home area. When an MS is roaming, registration, call origination and call delivery will take extra steps. Whenever data will be retrieved from the VLR, and the data are not available, then the VLR will send a message to the appropriate HLR to retrieve the data. The data consist of IMSI-to-MIN conversion, service profiles, shared secret data (SSD) for authentication, and other data needed to process calls. The most logical time to retrieve this data is when the MS registers with the system.

Once the data on a roaming MS are stored in the VLR, then call processing for any originating services (basic or supplementary) is identical to that of home MSs. However, there may be times when the MS originates a call before registration has been accomplished or when the VLR data are not available. At those times, an extra step will be added for the VLR to retrieve the data from the HLR. Thus any originating service has two optional steps where the VLR sends a message (using IS-41 signaling over SS7) to the HLR requesting data on the roaming MS. The HLR will return a message with the proper call information.

Call delivery is not possible to an unregistered MS because the network does not know where the MS is located. When the MS is registered with a system, call delivery to the roaming MS is possible. This section will discuss call delivery to roaming MSs in detail.

There are two cases of call delivery to roaming MS:

1. The MS has a geographic-based directory number (indistinguishable from a wireline number) and
2. The MS has a nongeographic number.

The call flows for both operations will be described.

When the MS has a geographic number, the MSC is assigned a block of numbers that are within the local numbering plan for the area of the world where the MSC is located. Call routing to the MS is then done according to the procedures for that of a wireline telephone.[2] If an MS associated with a MSC is not in its home area, the MSC will query the HLR for the location of the MS. The MSC then invokes call forwarding to the MSC where the MS is located, and the connection is made to the second MSC where call-terminating services are delivered according to the procedures in section 7.2.1.3. This procedure is inefficient because it results in two sets of network connections: originating switch to home MSC and home MSC to visited MSC.

Call delivery to a roaming MS is a cooperative effort among the home and visited MSC, the VLR and HLR, and the radio system. The detailed call flow steps for call delivery to a roaming MS with a geographic directory number follow (see fig. 7.6 for the call flow diagram):

1. A user in the world-wide phone network (wired or wireless) dials the directory number of the MS.

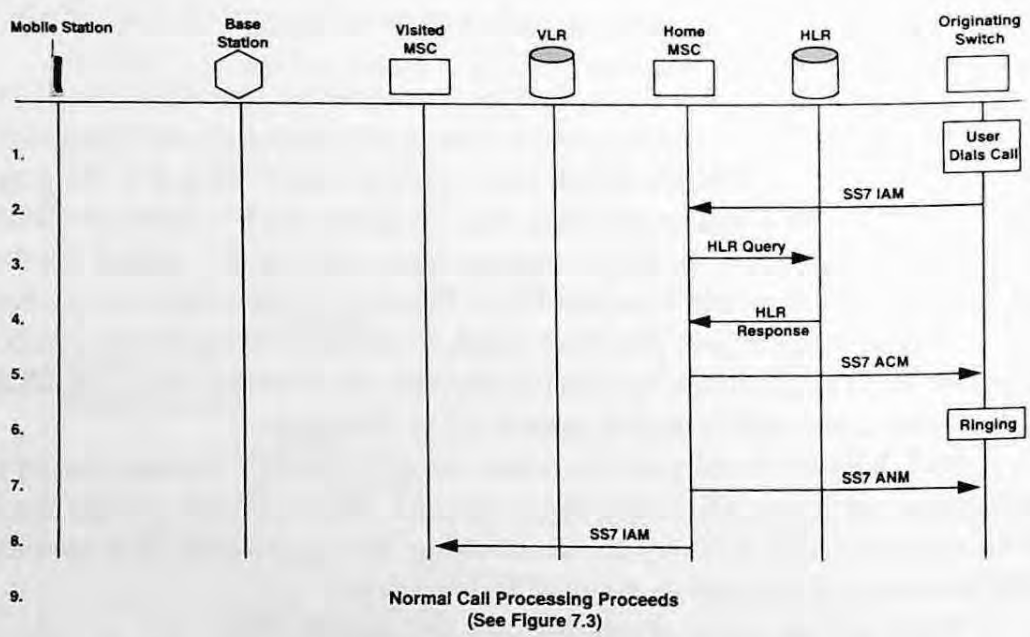


Figure 7.6 Call termination to a roaming mobile station with a geographic number.

2. For example, in New Jersey, 908-313-XXXX is used by the local cellular provider for cellular phones in Elizabeth, NJ. The wireline network routes calls to those numbers in a normal fashion and calls terminate on the cellular switch.

2. The originating switch sends an SS7 IAM to the home MSC.
3. The home MSC queries the HLR for the location of the MS.
4. The HLR returns the location of the visited system.
5. The home MSC sends an SS7 ACM message to the originating switch.
6. Ring is applied.
7. The home MSC sends an SS7 answer message to the originating switch.
8. The MSC invokes call forwarding to the MSC in the visited system and the forwarding (home) MSC switch sends an SS7 IAM to the visited MSC.
9. Call processing proceeds at step 3 of the terminating call flow.

When the MS has a nongeographic number, calls can be directed from an originating switch directly to the visited switch. Call delivery to a nongeographic number requires that the originating switch recognize the number as a nongeographic number and do special call processing for routing. This special processing is known as intelligent network (IN) processing. If the originating switch does not support IN, then it will route the call to a switch that supports IN. With IN support, the originating switch will recognize the nongeographic number and send an SS7 message to the HLR with a request for the location of the MS. The HLR will return a temporary directory number (on the visited MSC) that can be used to route to the MS in the visited system. Calls then proceed according to normal terminating call flows.

Call delivery to a roaming MS, with a nongeographic number is, therefore, a cooperative effort among the visited MSC, the VLR and HLR, and the radio system. The detailed call flow steps for call delivery to a roaming MS with a nongeographic directory number follow (see fig. 7.7 for the call flow diagram):

1. A user in the world-wide phone network (wired or wireless) dials the directory number of the MS.
2. The originating switch recognizes the number as a nongeographic number and sends an SS7 Query message to the HLR at the home MSC.
3. The HLR returns the location of the visited system with a directory number to use for further call processing.
4. The originating switch sends an SS7 IAM to the visited MSC.
5. Call processing proceeds at step 3 of the terminating call flow.

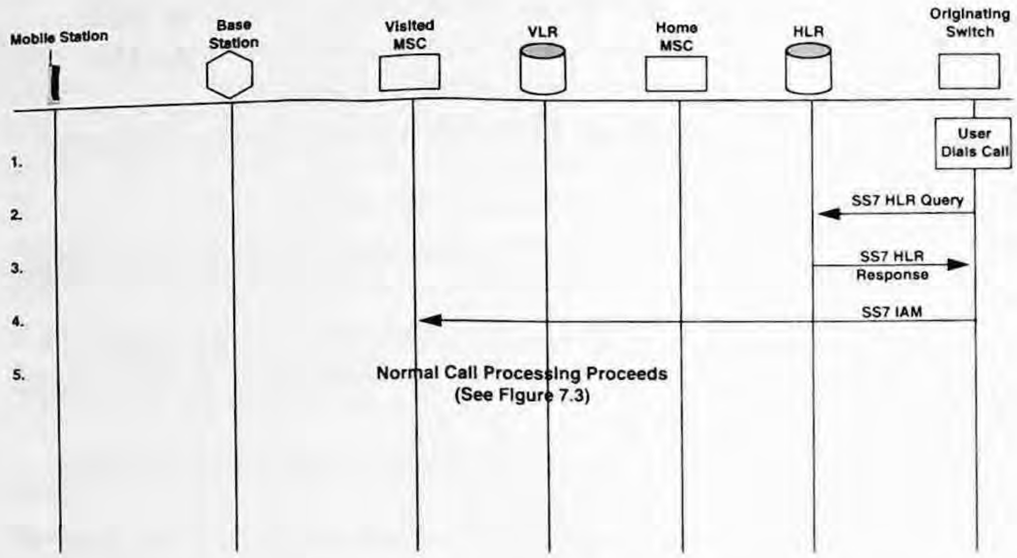


Figure 7.7 Call termination to a roaming mobile station with a nongeographic number.

7.2.2 Unique Challenge

There are several fraud problems with wireless phones [8, 9, 10] and the standards support security features of global challenge, SSD update, and unique challenge. We discuss the unique challenge here, and you can consult the standards for a description of other security features.

The unique challenge protects the network from fraudulent use by illegal mobile stations. At various times throughout a call, the network may want to challenge the validity of a mobile station communicating with the network. If the radio link communications are encrypted, it is unlikely that someone may have stolen the radio link from a legitimate user. The stealing of the radio link is called hijacking the link. Only those systems that operate unencrypted or have encryption disabled because of system overloads, national emergencies, or other reasons are subject to hijacking from illegal mobile stations.

The unique challenge can be sent to a MS at any time. It is typically initiated by the MSC in response to some event (registration failure and after a successful handoff are the most typical cases). The following steps are the call flow for a unique challenge (see fig. 7.8):

1. The MSC decides to perform a unique challenge.
2. The MSC sends a PCSAP message to the base station with TMSI (or MIN or IMSI if the MS is not registered) and RANDU.

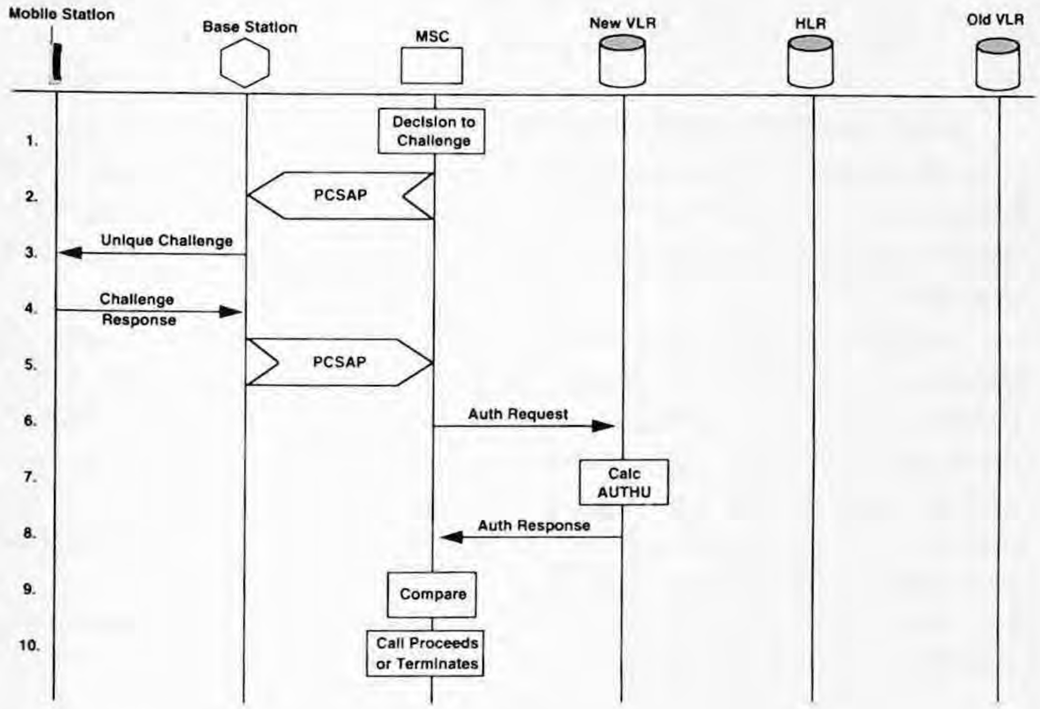


Figure 7.8 Shared secret key unique challenge.

3. The base station forwards the unique challenge message to the MS.
4. The mobile station calculates its specific response to the unique challenge (AUTHU) and sends to the base station a Unique Challenge Order Response message that includes TMSI (or MIN or IMSI), AUTHU, and other data as needed.
5. The base station forwards the message to the MSC in the PCSAP message.
6. The MSC sends an Authentication Request message to the VLR with TMSI (or MIN or IMSI), RAND, and AUTHU and requests that the VLR perform the same calculation as done by the MS.
7. The VLR checks its data base for TMSI (or MIN or IMSI). If the data are not in the VLR, the VLR queries the HLR for the data. When the data are in the data base at the VLR, it calculates the value of AUTHU.
8. The VLR returns a message to the MSC.
9. The MSC compares the AUTHU from the MS and from the VLR.
10. The MSC decides to continue or interrupt call processing. If the two AUTHUs match, then the MSC continues call processing. If

they do not match, then the MSC optionally may take action (e.g., terminate a call in progress or deregister the MS).

7.2.3 Supplementary Services

IS-41 supports several supplementary services (see section 4.3.2); however, only the call flow for call waiting is described herein. For other services, see either the standards or *Wireless and Personal Communications Systems* [9].

Call waiting provides notification to a wireless subscriber of an incoming call while the user's mobile station is in the busy state. Subsequently, the user can either answer or ignore the incoming call. Once the call is answered, the user can switch between the calls until one or more parties hang up. When either distant party hangs up, then the call reverts to a normal (non-call-waiting call). If the MS user hangs up, then both calls are cleared according to normal call-clearing functions.

The detailed call flow steps for call-waiting delivery to a mobile station follow (see fig. 7.9 for the call flow diagram):

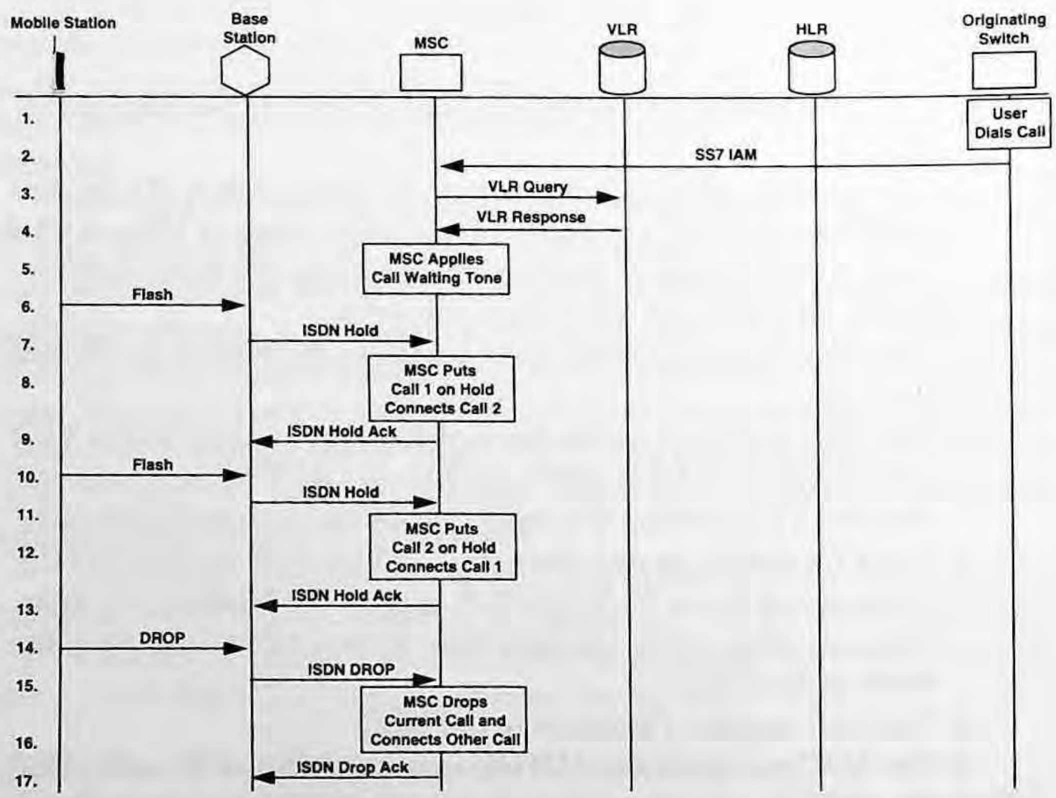


Figure 7.9 Call waiting.

1. User dials a call.
2. The originating switch sends an SS7 IAM to the MSC.
3. The MSC queries the VLR.
4. The VLR returns with a location of the MS that is within the serving system. If it is not, then the call is forwarded to the serving MSC.
5. The MSC determines that the MS is busy and subscribes to call waiting and thus applies a call-waiting tone.
6. The user presses the flash button (it may be SEND on some MSs) to answer the call-waiting indication, and the MS sends a Flash message to the base station.
7. The base station sends and ISDN Hold message to the MSC.
8. The MSC puts the first call on hold and connects the second call.
9. The MSC sends an ISDN Hold Acknowledge to the base station.
10. The User presses the flash button (it may be SEND on some MSs) to talk to caller 1, and the MS sends a Flash message to the base station.
11. The base station sends an ISDN Hold message to the MSC.
12. The MSC puts the second call on hold and connects the first call.
13. The MSC sends an ISDN Hold Acknowledge to the base station.
14. The user wants to drop the current call (either 1 or 2) and pushes the drop (or END) key and the MS sends a DROP message to the base station.
15. The base station sends an ISDN Drop message to the MSC.
16. The MSC drops the current call and connects the other call (the one currently on hold).
17. The MSC sends an ISDN Drop Acknowledge message to the base station.

7.2.4 Handoffs

A wireless telephone (mobile station) moves around a geographic area. When the station is idle, it periodically reregisters with the system according to the parameters described in section 7.2.1.1. When a call is active, the combination of the mobile station, the base station, and the MSC manage the communications between the base station and mobile station so that good radio link performance is maintained. The process whereby a mobile station moves to a new traffic channel is called *handoff*. The original analog cellular system processed handoffs by commanding the mobile station to tune to a new frequency. For analog cellular, the handoff process caused a short break in the voice path and a noticeable "click" was heard by both parties in the telephone call. For data modems, the click often causes data errors or loss of data synchronization.

For the CDMA (and wideband CDMA) systems, the characteristics of the spread spectrum communications permit the system to receive the mobile transmissions on two or more base stations simultaneously. In addition, the mobile station can simultaneously receive the transmissions of two or more base stations. With these capabilities, it is possible to process a handoff from one base station to another, or from one antenna face to another on the same base station, without any perceptible disturbance in the voice or data communications.

During handoff, the signaling and voice information from multiple base stations must be combined (or bridged) in a common point with decisions made on the "quality" of the data. Similarly, voice and signaling information must be sent to multiple base stations, and the mobile station must combine the results. The common point could be anywhere in the network but is typically at the MSC. The call flows described here for handoff assume that the MSC contains the bridging circuitry.

The CDMA system defines several types of handoffs.

- **Soft Handoff** occurs when the new base station begins communications with the mobile station while the mobile station is still communicating with the old base station. The network (MSC) combines the received signals from both base stations to process an uninterrupted signal to the distant party. The mobile station will receive the transmissions from the two base stations as additional multipaths in the RAKE receiver and will process them as one signal.
- **Softer Handoff** occurs when the mobile station is in handoff between two different sectors at the same base station. Typically, a base station is designed so that an antenna transmits and receives over a 60° or 120° sector rather than a full 360°. For full (360°) coverage, multiple base station antennas are then needed. For the purposes of discussion of softer handoffs, it is useful to designate a sector as a primary sector (i.e., the oldest sector serving the call). Since an MS will typically communicate only with three base stations during a soft handoff, only one (or none) softer handoff can be associated with a call at any particular time.
- **Hard Handoff** occurs when the two base stations are not synchronized or are not on the same frequency and an interruption in voice or data communications occurs. Hard handoffs can occur when

more than one frequency band is used, or the two base stations are not synchronized (e.g., there are in two different systems).

Another type of hard handoff occurs when there is no serving CDMA base station available and the mobile station must be directed to an analog cellular channel. In this book, we are discussing digital transmissions; however, during the transition time from analog to digital, there will be mixed systems in existence, and some mobile stations may be capable of both digital and analog operation. For more details, consult chapter 4, and the standards [4, 5, 6].

- **Semisoft Handoff** occurs when the handoff appears as a soft handoff within the network but the mobile station processes it as a hard handoff.

In CDMA, both the base station and the mobile station monitor the performance of the radio link and can request handoffs. Handoffs requested by a mobile station are called *mobile-assisted handoffs,* and those requested by the base station are called *base station-assisted handoffs*. Either side can initiate the handoff process whenever the following triggers occur:

- **Base Station Traffic Load.** The network can monitor loads at all base stations and trigger handoffs to balance loads between them to achieve higher traffic efficiency.
- **Distance Limits Exceeded.** Since all base stations and mobile stations are synchronized, both sides can determine base to mobile range. When the distance limit is exceeded, either side can request a handoff.
- **Pilot Signal Strength Below Threshold.** When the received signal strength of the pilot signal falls below a threshold, either side can initiate a handoff.
- **Power Level Exceeded.** When the base station commands a mobile station to increase its power and the maximum power level of the mobile station is exceeded, then either side can request a handoff.

The mobile station determines the parameters for the handoff request from the system parameters message in the CDMA system and the broadcast message in the wideband CDMA system. Both messages are transmitted on their systems' paging channels.

As we have described, the handoff process is a cooperative effort among the old and new base stations, the mobile station, and the MSC. The following call flows are based on a frame relay A-interface [12] between the base station and the MSC. The call flows are included as representative calls flows. The actual calls flows may be either standard or proprietary to an equipment vendor.

The detailed call flow steps for a CDMA soft handoff (beginning) follow (see fig. 7.10 for the call flow diagram):

1. The mobile station determines that another base station has a sufficient pilot signal to be a target for handoff.
2. The mobile station sends a Pilot Strength Measurement message to the serving base station.
3. The serving base station sends an Interbase Station Handoff Request message to the MSC.
4. The MSC accepts the Handoff Request and sends an Interbase Station Handoff Request message to the target base station.

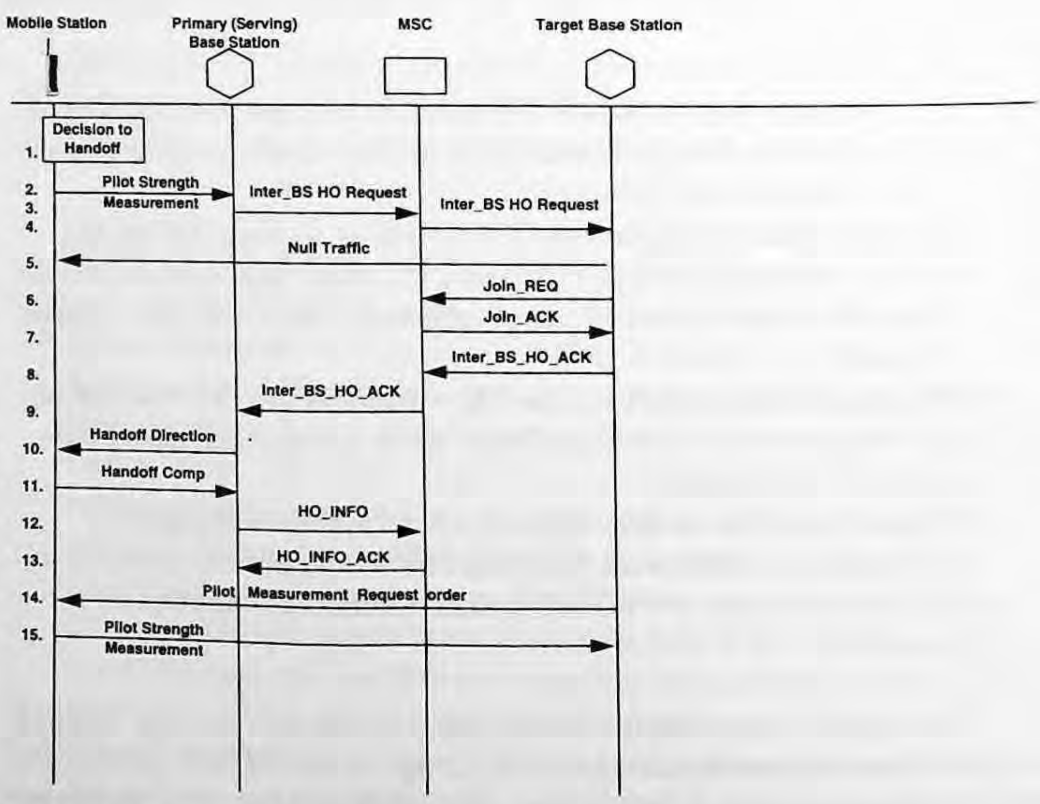


Figure 7.10 CDMA soft handoff—beginning.

5. The target base station establishes communication with the mobile station by sending it a Null Traffic message.
6. The target base station sends a Join Request message to the MSC.
7. The MSC conferences the connections from the two base stations so that the handoff can be processed without a break in the connection (i.e., soft handoff) and sends a Join Acknowlege message to the target base station.
8. The target base station sends an Interbase Station Handoff Acknowledgment message to the MSC.
9. The MSC sends a Interbase Station Handoff Acknowledgment message to the serving base station.
10. The serving base station sends a Handoff Direction message to the mobile station.
11. The mobile station sends a Handoff Complete message to the serving base station.
12. The new serving station sends a Handoff Information message to the MSC.
13. The MSC confirms the message with a Handoff Information Acknowledgment message.
14. The target base station sends a Pilot Measurement Request Order to the mobile station.
15. The mobile station sends a Pilot Strength Measurement message to the target base station.

The mobile unit is now communicating with two base stations (i.e., it is in soft handoff). Both base stations must communicate with the MSC. MSC uses the highest quality signals from the two base stations and sends transmitted signals to both base stations.

After the mobile station is in soft handoff, one of the signals may fall below a predetermined threshold (based on information sent in overhead messages on the control channel), and the mobile stations will request that one base station be removed from the connection. The detailed call flow steps for a CDMA soft handoff with the serving base station dropping off follow (see fig. 7.11 for the call flow diagram):

1. The mobile station determines that the serving base station has insufficient pilot signal to continue to be a base station in the soft handoff.

2. The mobile station sends a Pilot Strength message to the serving base station. The message requests that the base station drop off from the handoff.
3. The serving base station sends a Handoff Direction message to the mobile station. The message indicates which base station to be dropped from the soft handoff (in this case, the serving base station).
4. The mobile station sends a Handoff Complete message to the serving base station.
5. The serving base station sends an Interface Primary Transfer message to the target base station with relevant call record information.
6. The target base station confirms the message with an Interface Primary Transfer Acknowledge message.
7. The target base station then sends a Handoff Information message to the MSC.
8. The MSC sends a Handoff Information Acknowlege message to the target base station.

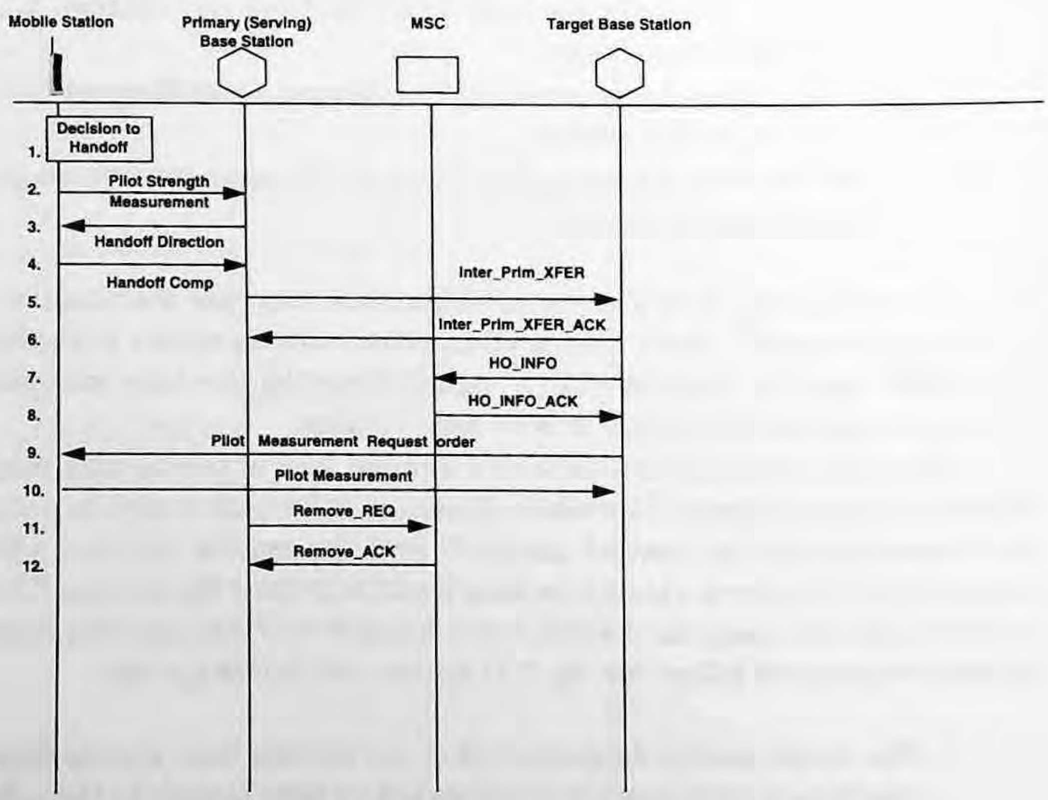


Figure 7.11 CDMA soft handoff: serving base station dropping off.

9. The target base station sends a Pilot Measurement Request Order to the mobile station.
10. The mobile station sends a Pilot Strength Measurement message to the target base station.
11. The old serving base station sends to the MSC a Remove Request message, which requests that the base station be dropped from the connection.
12. The MSC confirms the message by sending a Remove Acknowledge message to the old base station.

The mobile station is now communicating with the target base station (new serving base station). If additional soft handoffs are needed, the handoff beginning procedure is repeated.

The procedures to drop a target base station from a soft handoff are similar to those that drop the serving base station. The detailed call flow steps for a CDMA soft handoff with the target base station dropping off follow (see fig. 7.12 for the call flow diagram):

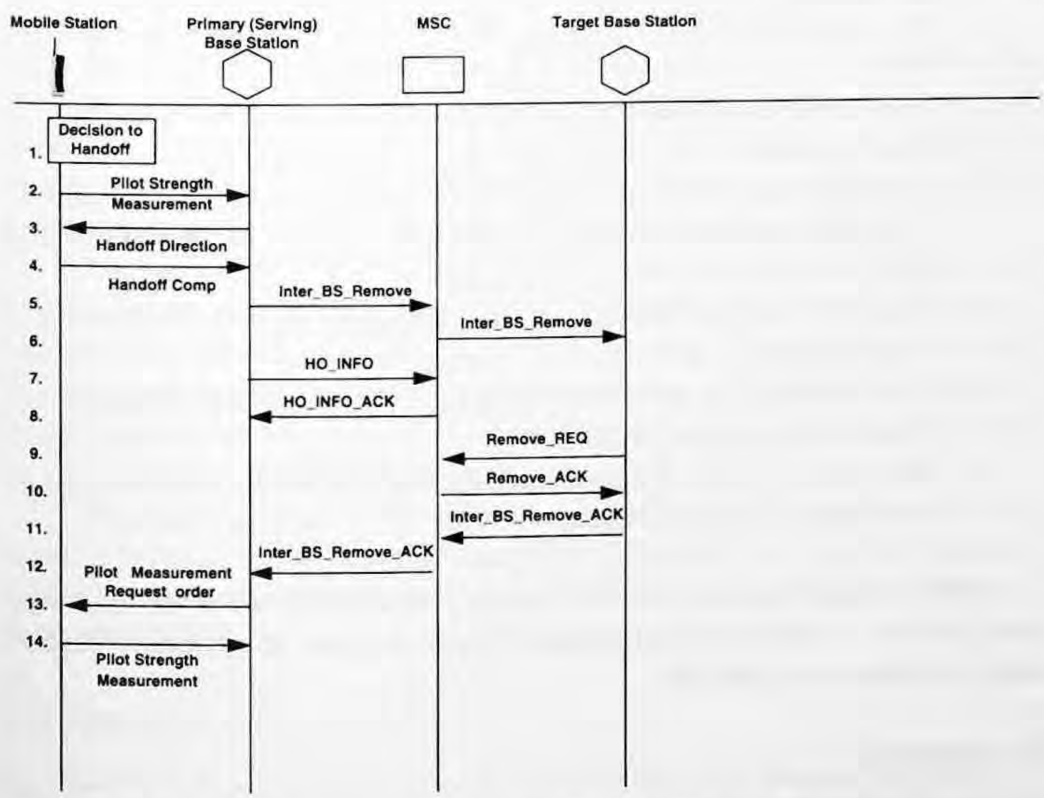


Figure 7.12 CDMA soft handoff: target base station dropping off.

1. The mobile station determines that the target base station has insufficient pilot signal to continue to be a base station in the soft handoff.
2. The mobile station sends a Pilot Strength message to the serving base station. The message requests that the target base station drop off from the handoff.
3. The serving base station sends a Handoff Direction message to the mobile station, which indicates the base station is to be dropped from the soft handoff (in this case, the target base station).
4. The mobile station sends a Handoff Complete message to the serving base station.
5. The serving base station sends an Interbase Station Remove message to the MSC.
6. The MSC sends an Interbase Station Remove message to the appropriate base station (in this case, the target base station).
7. The serving base station then sends a Handoff Information message to the MSC.
8. The MSC sends a Handoff Information Acknowlege message to the serving base station.
9. The target base station sends a Remove Request message to the MSC.
10. The MSC sends a Remove Acknowledge message to the target base station.
11. After the target base station removes its resource from the call, it sends an Interbase station Remove Acknowledge message to the MSC.
12. The MSC sends a Remove Acknowledge message to the serving base station.
13. The serving base station sends a Pilot Measurement Request Order to the mobile station.
14. The mobile station sends a Pilot Strength Measurement message to the serving base station.

The mobile station is now communicating only with the serving base station. If additional soft handoffs are needed, the handoff beginning procedure is repeated.

7.3 Summary

In this chapter, we have discussed the signaling applications for a CDMA wireless telephony system. Since end-to-end call flows are not presented in any of the standards but are distributed across several standards, we described several basic and supplementary call flows. First we examined the registration process. Because an unregistered phone cannot place or receive calls, it is necessary for a mobile station to register on the network. We described the call flow for registration including the authentication procedures used to validate the identity of the mobile station. After a phone is registered, it can place or receive calls; therefore, we described the call flows for a mobile-originated call and a mobile-terminated call. When a call is in progress, either the mobile station or the far end can end the call and release the connection. We examined the call flows for both release procedures.

An important component of wireless services is the ability to find and place calls to a roaming mobile station. Even though most mobile stations have a geographic number now, many will have nongeographic numbers in the future, so we described call flows for both. Geographic numbers are phone numbers that are located to a specific point on the world-wide phone system. Nongeographic numbers do not have a location associated with them, and the network must maintain a data base of the location of the phone. Additional routing steps are necessary to place a call to a mobile station with a nongeographic number.

Because fraud is a problem in the analog AMPS system, CDMA and W-CDMA implement cryptographic methods for combating fraud. At any time during call processing, the MSC can present a unique challenge to a mobile station to confirm its identity. We describe the call flow without revealing details that would permit the procedure to be defeated.

Even though cellular and personal communications systems (and CDMA, in particular) have a rich set of supplementary features, we examine the most common feature of call waiting. The various standards describe additional procedures for all the basic and supplementary services described in chapter 4. We encourage you to consult the standards [1–7, 11, 12] for additional information.

Finally, since the CDMA system processes handoffs differently than analog cellular or TDMA cellular/PCS systems, we describe the soft handoff process for CDMA and present call flows for soft handoff beginning and ending.

7.4 References

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CHAPTER 8

Voice Applications in the CDMA System

8.1 Introduction

This chapter discusses the voice-coding application in the CDMA system. The data application is discussed in chapter 10. Since voice encoding is critical to digital transmission and is the primary application that most wireless phone users need, we discuss the voice-coding algorithms used by both CDMA and wideband CDMA. CDMA uses an 8-kbps (or a 13.2-kbps) data rate for voice transmission, whereas wideband CDMA uses either ADPCM at 32 kbps or PCM at 64 kbps to provide a service more closely modeled on the current wireline network.

8.2 Voice Encoding

The wireline network is based on sending voice using digital pulse code modulation (PCM) at 64 kbps and sending data at rates of 64 kbps or higher multiples of 64 kbps. Many older analog facilities still exist, especially in residential areas, and use voice band modems at rates up to 28.8 kbps for data and analog electrical signals for voice. At the central office, the analog voice and analog data are converted to digital signals using PCM or, optionally, using modem pools for data.

It would be optimal if the identical systems could be used for wireless communications. Unfortunately the error rates on the radio channels are many orders of magnitude higher than those of the copper or fiber optic cables. In addition, PCM is inefficient for use over scarce and expensive radio channels.

Therefore, both the CDMA and wideband CDMA systems use some efficient method of voice coding and extensive error recovery techniques to overcome the harsh nature of the radio channel. The CDMA system uses a code-excited linear prediction (CELP) voice-encoding system at 8 kbps and optionally at 13 kbps; the wideband CDMA system uses ADPCM at 32 kbps as its primary voice-coding system and PCM at 64 kbps as an option.

In this section we will cover the various means used for both of these protocols to send voice signals over the radio channel.

8.2.1 Pulse Code Modulation

The simplest form of waveform coding scheme is linear PCM, in which the speech signal is band-limited, compressed, sampled, quantized, and encoded (see fig. 8.1). This approach is widely used for analog-to-digital conversion of a signal. In radio and telephone communications, it is not necessary to send the entire 20–20,000 Hz signal normally used for high-fidelity music. Intelligible speech communications can occur with a much narrower bandwidth and, therefore, more efficient range of frequencies. For telephone communications, the speech signal is band-limited to a frequency range 300–3300 Hz. To achieve telephone quality speech, 12 bits per sample are required at a sampling rate of 8000 samples per second. However, by using a logarithmic sampling system, 8 bits per sample are sufficient. Each sample is then quantized into one of 256 levels. Telephone speech uses two widely different variations of PCM to achieve quality speech (μ-law and A-law PCM). Both are based on a non-uniform quantization of the signal amplitude according to a logarithmic scale rather than a linear scale.

The decoder for PCM (fig. 8.2) inverts the stages of the encoding process. PCM encoding and decoding are inherently simple systems. However, they require a high bit rate for transmission.

For PCM, North America and Japan use μ-law encoding where the output digital signal $s(t)$ is related to the input signal $i(t)$ by

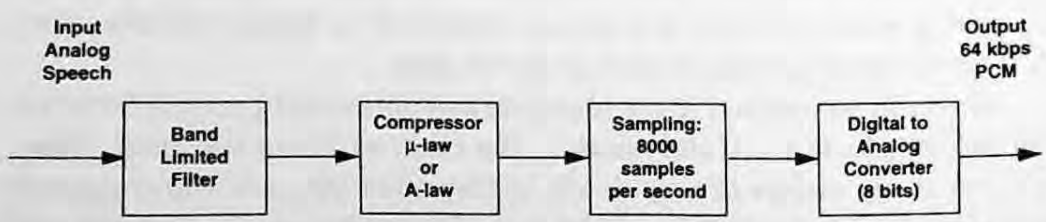

Figure 8.1 PCM encoder.

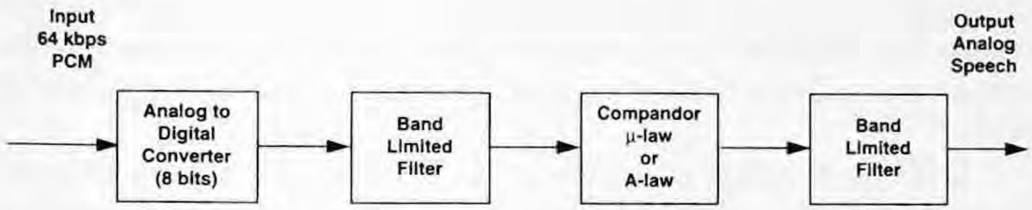

Figure 8.2 PCM decoder.

$$s(t) = \sin(i(t))\frac{\ln(1+\mu|i(t)|)}{\ln(1+\mu)}, \qquad -1 \le i(t) \le 1 \tag{8.1}$$

where a typical value for $\mu = 255$ is used in the United States.

In equation (8.1), the input signal is normalized to a range of ±1. It can be noted that for small $i(t)$, $s(t)$ approaches a linear function and that for large $i(t)$, $s(t)$ approaches a logarithmic function. The purpose of the μ-law encoding is to improve the signal-to-noise ratio for weak speech signals. The overall data rate is 64 kbps with sampling at 8 kbps and 8 bits per sample.

In Europe, PCM uses A-law encoding where the output digital signal $s(t)$ is related to the input signal $i(t)$ by

$$\begin{aligned} s(t) &= \sin(i(t))\frac{1+\ln(A|i(t)|)}{1+\ln A} & -\frac{1}{A} \le |i(t)| \le 1 \\ s(t) &= \sin(i(t))\frac{1+(A|i(t)|)}{1+\ln A} & 0 \le |i(t)| \le \frac{1}{A} \end{aligned} \tag{8.2}$$

Where a typical value of $A = 87.6$ is used in Europe.

In equation (8.2), the input signal is also normalized to a range of ±1. Note that $s(t)$ is logarithmic for $|i(t)| < 1/A$ and linear for $|i(t)| > 1/A$. Thus, A-law provides a somewhat flatter signal-to-distortion performance compared to μ-law when the signal is greater than $1/A$, at the expense of poorer performance at low signal levels.

Telephone communications that cross continental borders must have conversion routines in their transmission paths if the two continents use different encoding laws.

8.2.2 Adaptive Differential Pulse Code Modulation

High bit rates are not attractive for wireless systems since the capacity of the system is low. Higher system capacities are obtained with differential coders, where compression can be applied dynamically, such as adaptive predictive coding (APC) and adaptive differential pulse code modulation (ADPCM). The reason for these coders is to achieve a better signal-to-quantization noise performance and a lower coding rate over PCM.

Differential coders generate error signals, as the difference between the input speech samples and corresponding prediction estimates. The error signals are quantized and transmitted. ADPCM and APC differential coders are often used for intermediate bit rate between 16 and 32 kbps.

ADPCM employs a short-term predictor that models the speech spectral envelope. It achieves network-quality speech (mean opinion score [MOS] of 4.1 or better) at 32 kbps. This is a low-complexity, low-delay coder of reasonable robustness with channel bit error rates in the range of 10^{-3} to 10^{-2}. The ADPCM coder is well suited for wireless access applications.

In an ADPCM encoder (fig. 8.3), first the analog speech is converted to PCM. If the signal is already PCM, from the network for example, then the analog to the PCM step is not needed. The A-law or μ-law encoded signal is then converted to a uniform PCM level (i.e., equal steps between levels) signal. The encoder generates a difference signal between the converted signal and an estimated signal and encodes the estimated signal using 15 levels. The resultant signal is transmitted at 32 kbps (half the rate for PCM). In the encoder, the signal estimator is generated by an inverse quantizer and an adaptive predictor. The use of differential signals and proper design of the predictor enables an overall coding efficiency improvement over PCM.

In the ADPCM decoder (fig. 8.4), the input 32-kbps signal is processed by a inverse adaptive quantizer and an adaptive predictor. The

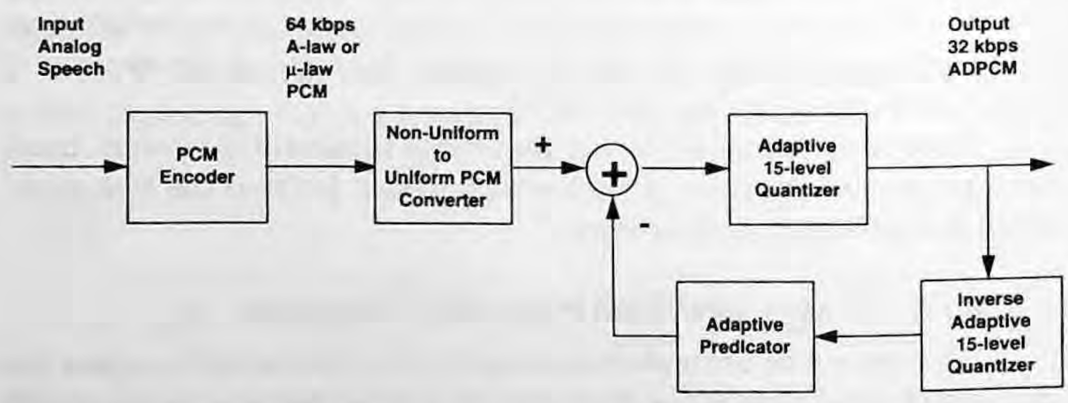

Figure 8.3 ADPCM encoder. (Reproduced under written permission of the copyright holder [TIA].)

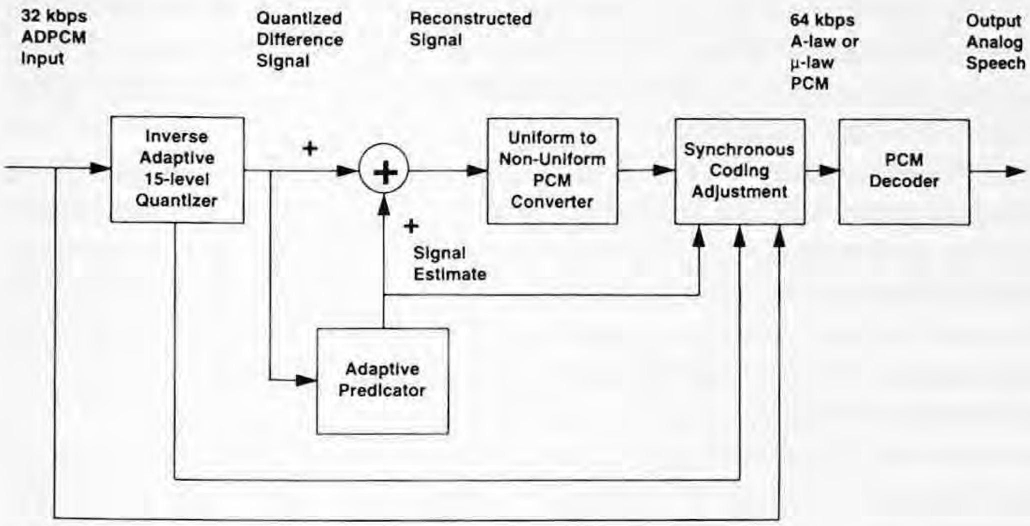

Figure 8.4 ADPCM decoder. (Reproduced under written permission of the copyright holder [TIA].)

output of the quantizer and the predictor are combined to generate a reconstructed signal that is converted back to PCM. The regenerated PCM signal is then processed (in the synchronous coding adjustment stage) with signals from the input, the quantizer output, and the predictor output to generate the A-law or μ-law PCM signal. The processing in the synchronous coding adjustment stage ensures that the PCM signal is correctly modeled by converting it to uniform PCM and comparing the error signals that result from the actual received signal. If an error occurs, it is corrected before the output PCM signal is generated. The ADPCM signal conforms to CCITT recommendation G.721. Finally, if analog speech is needed, the signal is processed by a PCM decoder.

The IS-665 standard supports a modified version of ADPCM that has been optimized for the PCS environment. The functional operation is the same as ADPCM; the coding algorithm has been modified. Refer to the IS-665 standard for the details.

8.2.3 Code-Excited Linear Predictor

PCM, ADPCM, and APC operate in the time domain. No attempt is made to understand or analyze the information that is being sent. To achieve lower coding rates, redundancy removal techniques operating in the frequency domain have been used successfully. Frequency domain waveform coding algorithms decompose the input speech signal into sinusoidal components with varying amplitudes and frequencies. Thus, the speech is modeled as a time-varying line spectrum. Frequency domain coders are systems of moderate complexity and operate well at a medium bit rate (16 kbps). When designed to operate in the range of 4.8–9.6 kbps, the complexity of the approach used to model the speech spectrum increases considerably.

The other class of speech-coding techniques consists of algorithms called *vocoders*, which attempt to describe the speech production mechanism in terms of a few independent parameters serving as the information-bearing signals. These parameters attempt to model the creation of the voice by the vocal tract, decompose the information, and send it to the receiver. The receiver attempts to model an electronic vocal tract to produce the speech output.

The modeling operates in the following way. Vocoders consider that speech is produced from a source-filter arrangement. Voiced speech is the result of exciting the filter with a periodic pulse train (simulating the opening and closing of the vocal cords). Unvoiced speech is the result of exciting the filter with random noise (simulating air rushing past a constriction in the vocal tract). Vocoders operate on the input signal using an analysis process based on a particular speech production model and extract a set of source-filter parameters that are encoded and transmitted. At the receiver, they are decoded and used to control a speech synthesizer, which corresponds to the model used in the analysis process. Provided that all the perceptually significant parameters are extracted, the synthesized signal, as perceived by human ear, resembles the original speech signal. Nonspeech signals are often not modeled well, so this method works poorly for analog modems.

Vocoders are medium-complexity systems and operate at low bit rates, typically 2.4 kbps, with synthetic-quality speech. Their poor-quality speech is due to the oversimplified source model used to drive the filter and the assumption that the source and filter are linearly independent.

In the bit rates, from about 5 to 16 kbps, the best speech quality is obtained by using hybrid coders, which use suitable combinations of waveform-coding and vocoder techniques. A simple hybrid coding scheme for telephone-quality speech with a few integrated digital signal processors is the residual excited linear prediction (RELP) coding. This belongs to a class of coders known as an analysis-synthesis coder based on linear predictive coding (LPC).

The RELP systems employ short-term (and in certain cases, long-term) linear prediction to formulate a difference signal (residual) in a feed-forward manner. RELP systems are capable of producing communications quality speech at 8 kbps. These systems use either pitch-aligned, high-frequency regeneration procedures or full-band pitch prediction in time domain to remove the pitch information from the residual signal prior to band-limitation/decimation. At bit rates less than 9.6 kbps, the quality of the recovered speech signal can be improved significantly by an analysis by synthesis (AbS) optimization procedure to define the excitation signal. In these systems, both the filter and the excitation are defined on a short-term basis using a closed-loop optimization process that minimizes a perceptually weighted error measure formed between the input and decoded speech signals.

CDMA uses a variation of RELP called code-excited linear prediction. With this technique, the CELP decoder (fig. 8.5) uses a codebook to generate inputs to a synthesis filter. The codebook is characterized by its codebook index I and gain G. The spectral filter is characterized by three sets of parameters: the pitch spectral lines a, the pitch lag L, and the pitch gain b. The output of the filter is processed by a post filter and gain adjustment.

CDMA implements a rate 1 encoder at 8.55 kbps and supports rates of 4, 2, and 0.8 kbps (rates 1/2, 1/4 and 1/8, respectively). Each of the rates uses successively less bits for encoding the values of I, G, L, b, and a. At rate of 1/8 (fig. 8.6), insufficient bits are available to send the codebook index I, and a pseudorandom code generator (synchronized at both ends) is used and seeded by a random seed of value CBSEED.

The basic frame for CDMA is 20 ms. At rate 1, 160 bits are sent for encoding the data plus an 11-bit parity check field. Fewer bits are used at lower data rates (see table 8.1).

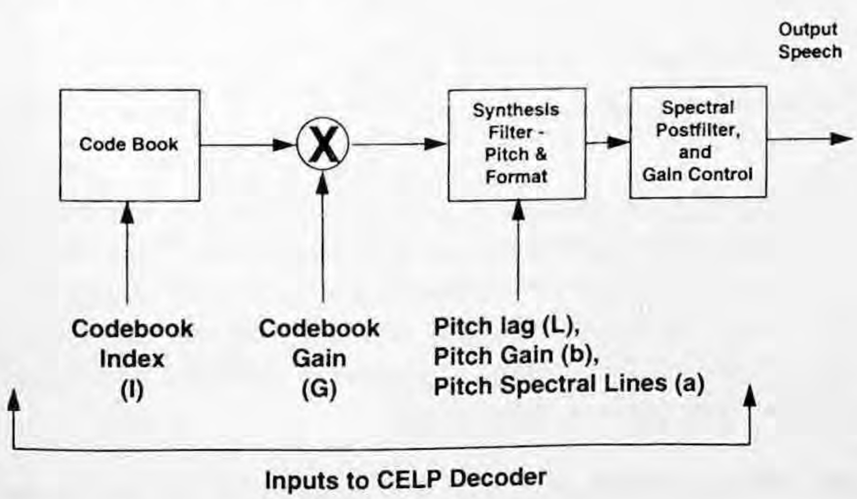

Figure 8.5 CELP decoder for rates 1, 1/2, and 1/4.

Table 8.1 CELP Parameters for Various Coding Rates

CELP Parameters	Rate 1	Rate 1/2	Rate 1/4	Rate 1/8
Line spectral pairs i bits	40	20	10	10
i updates per frame	1	1	1	1
Total i bits per frame	40	20	10	10
Pitch lag L bits	7	7	7	0
L updates per frame	4	2	1	0
Total L bits per frame	28	14	7	0
Pitch gain b bits	3	3	3	0
b updates per frame	4	2	1	0
Total b bits per frame	12	6	3	0
Codebook index I bits	7	7	7	0
I updates per frame	8	4	2	—
Total I bits per frame	56	28	14	0
Codebook gain G bits	3	3	3	2
G updates per frame	8	4	2	1
Total G bits per frame	24	12	6	2
Codebook seed CBSEED bits	0	0	0	4
CBSEED updates per frame	—	—	—	1
Total CBSEED bits	—	—	—	4
Parity check bits per frame	11	0	0	0
Total number of bits per frame	171	80	40	16

The CELP speech encoder requires three steps to implement. First the line spectral pairs (LSP) i values are determined. Then the LSP values are used in an analysis by synthesis process to determine the values for the pitch lag L and gain b. Finally, the values of i, L, and b are used in a second AbS step to determine the codebook indices I and gains G. We now describe these steps in more detail.

- **LSP Determination** (fig. 8.7). The encoder for the LSP codes first converts the speech to uniform PCM with at least 14 bits. If the

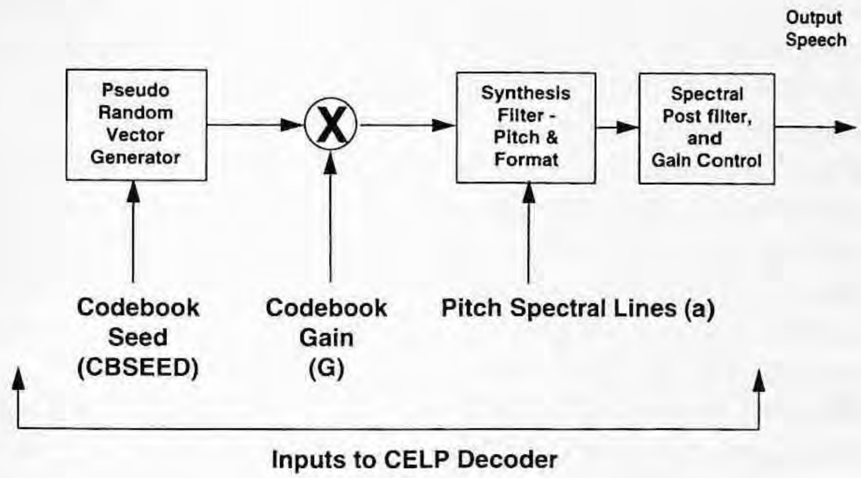

Figure 8.6 CELP decoder for rate of 1/8.

encoder is in a base station, then the received speech is most likely μ-law PCM; if the encoder is in a mobile station, then the received speech is analog. After the speech is converted to PCM, it is processed to remove the DC component and filtered by a Hamming window. The autocorrelation of the sampled output is then com-

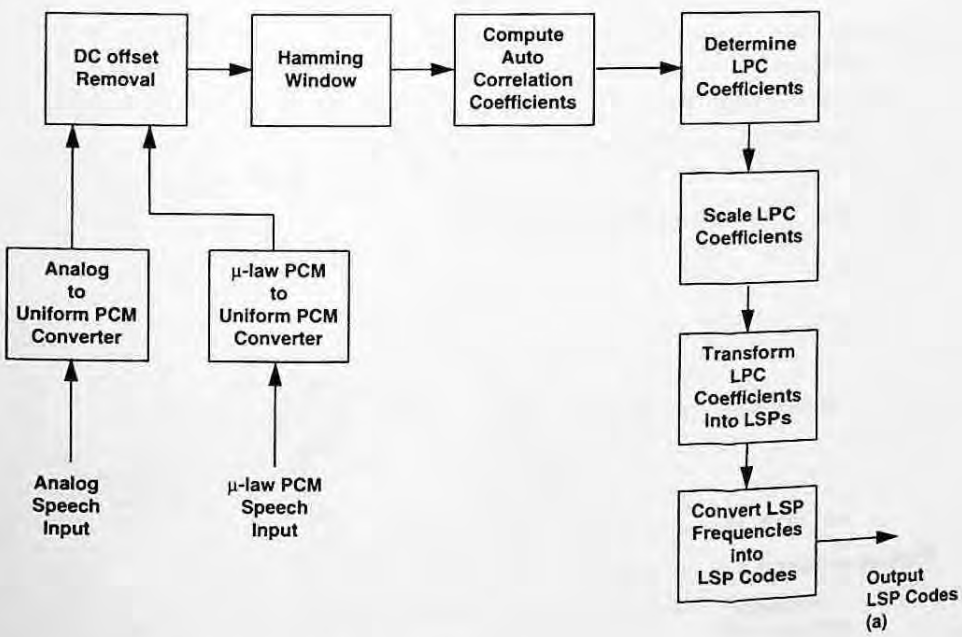

Figure 8.7 CELP encoder for LSP codes.

puted and used to determine the coefficients for the linear predictive coding. The LPC coefficients are then scaled, transformed into the frequency components, and converted into the values for the *i* bits of the coder output.

- **The Pitch Lag and Gain Bits** (fig. 8.8). They are computed by a recursive process where the output of the PCM encoder is combined with the LSP codes previously calculated and with all possible values of pitch and gain. For each value of pitch and gain, an error function is computed, and the transmitted values for pitch and gain are chosen to minimize the error.
- **The Codebook Index and Gain** (fig. 8.9). They are computed in a similar recursive process using the uniform PCM signal; the computed values for frequency, pitch, and gain; and all possible codebook values and gains.

For the rate of 1/8, codebook indices are not computed, but a random vector generated at both sides is used.

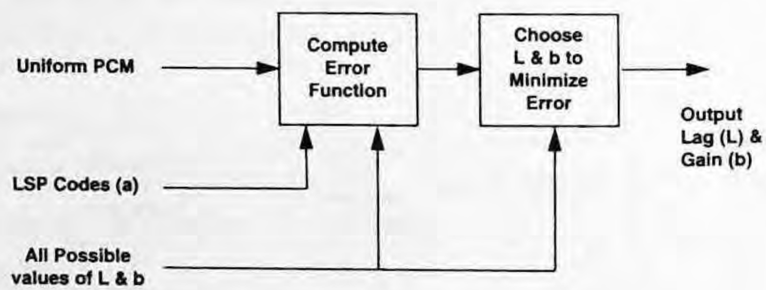

Figure 8.8 CELP encoder for pitch parameters.

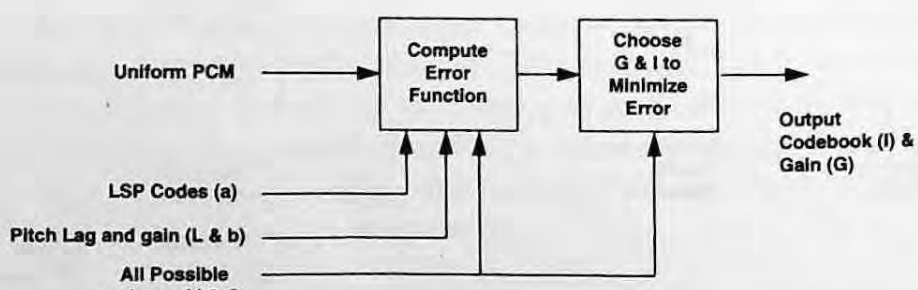

Figure 8.9 CELP encoder for codebook values.

8.3 Summary

This chapter discussed the digital voice-encoding systems used for both CDMA and wideband CDMA. Conventional wireline systems transmit voice by digitizing the voice signal using PCM at a rate of 64 kbps. While it is possible to use PCM in wireless systems (the W-CDMA system uses it as an option), the capacity of the wireless system is lower compared to using other digitizing methods for voice. The two CDMA systems use different approaches to digitizing the voice signal. The CDMA system uses a CELP at 8 or 13.2 kbps to digitize voice. CELP systems model the operation of the human vocal tract to code speech efficiently. We described the operation of the CELP encoder and decoder at a high level. The wideband CDMA system uses a version of PCM called adaptive differential PCM at 32 kbps. To understand ADPCM, we first described PCM and then the ADPCM system. The W-CDMA system uses a modified version of ADPCM. Throughout the descriptions on voice coding, our goal has been to explain the coding systems at a high level so that you can understand the operation of the system and read the standards with some understanding of the motivation for them. Those of you who need to design systems or want additional information are encouraged to read the standards.

8.4 References

1. TIA IS-96A, "Speech Service Option Standard for Wideband Spread Spectrum Digital Cellular System."
2. TIA IS-665, "W-CDMA (Wideband Code Division Multiple Access) Air Interface Compatibility Standard for 1.85–1.99 GHz PCS Applications."
3. Recommendation G162, CCITT Plenary Assembly, Geneva, May–June 1964, Blue Book, Vol. 111, P. 52.
4. ITU Recommendation G.711.

CHAPTER 9

RF Engineering and Facilities Engineering

9.1 Introduction

This chapter presents basic guidelines for engineering a CDMA system. The topic is extremely complex and cannot be covered extensively in a single chapter. Moreover, many of the effective techniques are currently being developed as CDMA is deployed in the commercial market. This chapter discusses several topics that are germaine to the engineering of a CDMA system: propagation models, link budgets, the transition from analog operation to CDMA operation, facilities engineering, radio link capacity, and border cells on an boundary between two service providers.

9.2 Radio Design for a Cellular/PCS Network

Many factors need to be considered early in the design of a cellular/PCS network for a metropolitan urban area. For example, the extent of radio coverage for indoor locations, the quality of service for different environments, efficient use of the spectrum, and the evolution of the network are some of the key factors that need to be carefully evaluated by all prospective service providers. Often, these factors are further complicated by the constraints imposed by the operating environments and regulatory issues. A system designer must carefully balance all the trade-offs to ensure that the network is robust, future-proof, and of high service quality.

9.2.1 Radio Link Design

For any wireless communications system, the first important step is to design the radio link. This is required to determine the base station density in different environments as well as the corresponding radio coverage. For a wireless network to provide good quality indoor and outdoor service in an urban environment, flexibility and resilience should be incorporated into the design. The transmit power of handsets will be the determining factor for a CDMA system with balanced uplink/downlink power.

Although the mobile antenna gain does not affect the balancing of the link budget, it is an important factor in the design of the power budget for handset coverage. From a user's point of view, a cellular/PCS network should imply that there is little restriction on making or receiving calls within a building or traveling in a vehicle using handsets. A system should be designed to allow the antenna of a handset to be placed in non-optimal positions. In addition, the antenna may not even be extended when calls are being made or received. In normal system designs, it is assumed that the gain of a mobile antenna is 0 dBi.[1] However, allowing for the handset antennas to be placed in suboptimal positions, a more conservative gain of –3 dBi should be used. In reality, the antenna gain, because of the positioning of an antenna in an arbitrary position or with the antenna retracted into the handset housing, could be as low as –6 to –8 dBi, depending on specific handsets and their corresponding housing designs.

9.2.2 Coverage Planning

The most important design objective of a cellular/PCS network is to provide a near-ubiquitous radio coverage. One of the most important considerations in the radio coverage planning process is the propagation model. The accuracy of the prediction by a particular model depends on its ability to account for the detailed terrain, vegetation, and buildings. This accuracy is of vital importance to determine the path loss and, hence, the cell sizes and the infrastructure requirement of a cellular/PCS network. An overestimation will lead to an inefficient use of the network resources, whereas an underestimation will result in poor radio coverage. Propagation models generally tend to oversimplify real-life propagation conditions and may be grossly inaccurate in complex metropolitan

1. dBi refers to the gain relative to an isotropic antenna.

urban environments. The empirical propagation models provide general guidelines only, and they are too simplistic for accurate network design. Accurate field measurements must be made to provide information on the radio coverage in an urban environment. Measured data can be used either directly in the planning process to access the feasibility of individual cell site or indirectly to calibrate the coefficients of the empirical propagation model to achieve better characterization of a specific environment.

Radio propagation in an urban environment is subject to shadowing. To ensure that the signal level in 90 percent of the cell area is equal to or above the specified threshold, a shadow-fading margin, which is dependent on the standard deviation of the signal level, must be included in the link budget. For a typical urban environment, a shadow-fading margin of 8–9 dB should be used based on the assumption that the path loss follows an inverse 2-5 exponent law, i.e., the path loss is inversely proportional to the distance of separation raised to a power between 2 and 5. The value of the power is dependent upon propagation characteristics.

Another critical factor that affects the radio coverage is the penetration loss for both buildings and vehicles. If the radio coverage for the outer portion of a building is sufficient, then an assumed penetration loss of 10–15 dB should be adequate. However, if calls are expected to be received and originated within the inner core of the building, a penetration loss of about 30 dB should be used. Similarly, for in-vehicle coverage, the penetration loss is equally important. A car could experience a penetration loss of 3–6 dB, whereas vans and buses have even larger variations. The penetration loss at the front of a van should be no more than that experienced in a car, but the loss at the back of a van could be as high as 10–12 dB, depending on the amount of window space. Thus, for design purposes, a high penetration loss should be assumed to ensure a good service quality. For an urban environment, as building penetration loss is the dominant factor, in-vehicle penetration will generally be sufficient as a consequence.

9.3 Propagation Models

Propagation models are used to determine how many base stations are required to provide the coverage requirements needed for the network. Initial network design typically engineers for coverage. Later growth of network design engineers for capacity. Some systems may need to start with wide area coverage and high capacity and, therefore, may start at a later stage of growth.

The coverage requirement is coupled with the traffic-loading requirements, which rely on the propagation model chosen to determine the traffic distribution and the off-loading from an existing base station to new base stations as part of a capacity relief program. The propagation model helps to determine where the base stations should be located to achieve an optimal position in the network. If the propagation model used is not effective in helping to place base stations correctly, the probability of incorrectly deploying a base station into the network is high.

The performance of the network is affected by the propagation model chosen because it is used for interference predictions. As an example, if the propagation model is inaccurate by 6 dB, then E_b/N_0 could be 13 or 1 dB (assuming that $E_b/N_0 = 7$ dB is the design requirement). Based on traffic-loading conditions, designing for a high E_b/N_0 level could negatively affect financial feasibility. On the other hand, designing for a low E_b/N_0 would degrade the quality of service.

The propagation model is also used in other system performance aspects including handoff optimization, power level adjustments, and antenna placements. Although no propagation model can account for all perturbations experienced in the real world, using one or more models for determining the path losses in the network is essential. Each of the propagation models being used in the industry has pros and cons. It is through a better understanding of the limitations of each of the models that a good RF engineering design can be achieved in a network.

9.3.1 Modeling for the Outside Environment

9.3.1.1 Analytical Model The propagation loss between the base station and the mobile station in the outside environment has been extensively studied. The propagation loss is generally expressed by the following expression [3,4]:

$$P(R) = N(R, \sigma) + n \log \frac{R}{R_0} \tag{9.1}$$

where $P(R)$ = loss at distance R relative to the loss at a reference distance R_0,
n = path loss exponent,
σ = standard deviation, typically 8 dB.

The second term on the right-hand side of equation (9.1) represents a constant attenuation in the outside environment between the base station and the mobile station. Typically, n approximately equals 4, although it may range between 2 (which equals the loss in free space) and 5. If n is equal to 4, then the signal will be attenuated 40 dB if the distance increases 10 times with respect to the reference distance. The first term in equation (9.1) represents the variation in the loss about the average path loss. This function is an approximate log-normal distribution with an average equal to the second term and a standard deviation of approximately 8 dB. It has been found that this value is applicable for a wide range of radio environments, including urban and rural areas.

9.3.1.2 Empirical Models Several empirical models have been suggested and used to predict propagation path losses. We discuss the two widely used models—the Hata-Okumura model and the Walfisch-Ikegami model.

The Hata-Okumura Model [1]. Most of the propagation tools use a variation of Hata's model. Hata's model is an empirical relation derived from the technical report made by Okumura [6] so that the results could be used in computational tools. Okumura's report consists of a series of charts that have been used in radio communication modeling. The following are the expressions used in Hata's model in order to determine the mean loss L_{50}:

Urban area:

$$\begin{aligned} L_{50} = {} & 69.55 + 26.16 \log f_c - 13.82 \log h_b - a(h_m) + \\ & (44.9 - 6.55 \log h_b) \log R \text{ dB} \end{aligned} \tag{9.2}$$

where f_c = frequency (MHz),
L_{50} = mean path loss (dB),
h_b = base station antenna height (m),
$a(h_m)$ = correction factor for mobile antenna height (dB),
R = distance from base station (km).

The range of the parameters for which Hata's model is valid is

$$150 \le f_c \le 1{,}500 \text{ MHz},$$

$$30 \le h_b \le 200 \text{ m},$$

$$1 \le h_m \le 10 \text{ m},$$

$$1 \le R \le 20 \text{ km}.$$

$a(h_m)$ is computed as:

For a small or medium-sized city:

$$a(h_m) = (1.1 \log f_c - 0.7)h_m - (1.56 \log f_c - 0.8) \text{ dB} \tag{9.3}$$

For a large city:

$$a(h_m) = 8.29(\log 1.54\, h_m)^2 - 1.1 \text{ dB}, \qquad f_c \le 200 \text{ MHz} \tag{9.4}$$

or

$$a(h_m) = 3.2(\log 11.75\, h_m)^2 - 4.97 \text{ dB}, \qquad f_c \ge 400 \text{ MHz} \tag{9.5}$$

Suburban Area:

$$L_{50} = L_{50}(urban) - 2\left[\left(\log\left(\frac{f_c}{28}\right)^2\right) - 5.4\right] \text{dB} \tag{9.6}$$

Open Area:

$$L_{50} = L_{50}(urban) - 4.78(\log f_c)^2 + 18.33 (\log f_c) - 40.94 \text{ dB} \tag{9.7}$$

Hata's model does not account for any of the path-specific correction used in Okumura's model.

Okumura's model [6] tends to average over some of the extreme situations and does not respond sufficiently quickly to rapid changes in the radio path profile. The distance-dependent behavior of Okumura's model is in agreement with the measured values. Okumura's measurements are valid only for the building types found in Tokyo. Experience with comparable measurements in the United States has shown that a "typical" United States suburban situation is often somewhere between Okumura's suburban and open areas. Okumura's suburban definition is more representative of residential metropolitan area with large groups of "row" houses.

Okumura's model requires that considerable engineering judgment be used, particularly in the selection of the appropriate environmental factors. Data are needed in order to be able to predict the environmental factors from the physical properties of the buildings surrounding a mobile receiver. In addition to the appropriate environmental factors, path-specific corrections are required to convert Okumura's mean path loss predictions to the predictions that apply to the specific path under study. Okumura's techniques for correction of irregular terrain and other path-specific features require engineering interpretations and are thus not readily adaptable for computer use.

The Walfisch/Ikegami Model [12]. This model[2] is used to estimate the path loss in an urban environment for cellular communication. The model is a combination of the empirical and deterministic model for estimating the path loss in an urban environment over the frequency range of 800–2000 MHz. This model is used primarily in Europe for the GSM system and in some propagation models in the United States. The model contains three elements: free space loss, roof-to-street diffraction and scatter loss (see fig. 9.1), and multiscreen loss. The expressions used in this model are

$$L_{50} = L_f + L_{rts} + L_{ms} \tag{9.8}$$

or

$$L_{50} = L_f \text{ when } L_{rts} + L_{ms} \le 0 \tag{9.9}$$

where L_f = free space loss,
L_{rts} = roof-to-street diffraction and scatter loss,
L_{ms} = multiscreen loss.

Free space loss is given as:

$$L_f = 32.4 + 20 \log R + 20 \log f_c \text{ dB} \tag{9.10}$$

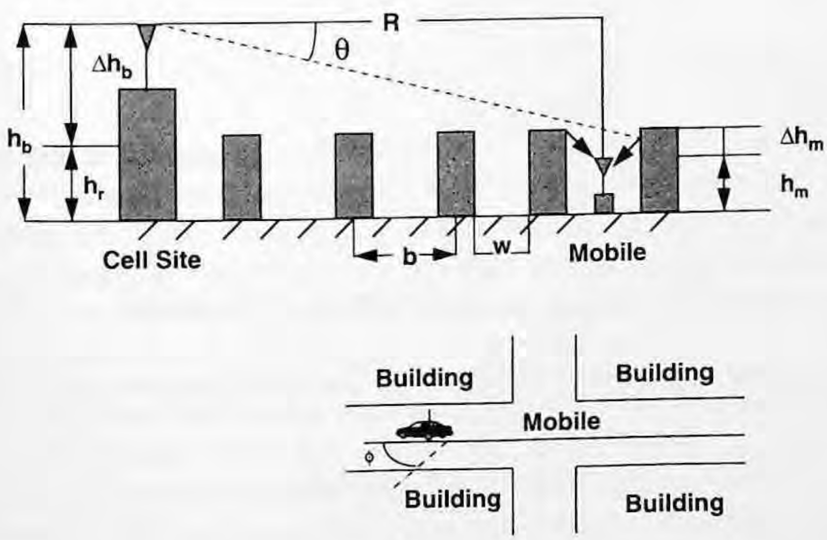

Figure 9.1 The Walfisch-Ikegami propagation model.

2. This model is also known as the Cost 231 model.

The roof-to-street diffraction and scatter loss is given as

$$L_{rts} = (-16.9) - 10 \log W + 10 \log f_c + 20 \log \Delta h_m + L_0 \text{ dB} \quad (9.11)$$

where W = street width (m),
$\Delta h_m = h_r - h_m$ (m).
$L_0 = -9.646$ dB $\quad 0 \le \phi \le 35$ degree
$L_0 = 2.5 + 0.075(\phi - 35)$ dB $\quad 35 \le \phi \le 55$ degree
$L_0 = 4 - 0.114(\phi - 55)$ dB $\quad 55 \le \phi \le 90$ degree
where ϕ = incident angle relative to the street

The multiscreen loss is given as

$$L_{ms} = L_{bsh} + k_a + k_d \log R + k_f \log f_c - 9 \log b \quad (9.12)$$

where b = distance between buildings along the radio path (m)
$L_{bsh} = -18 \log 11 + \Delta h_b \quad h_b > h_r,$
$L_{bsh} = 0 \quad h_b < h_r,$
$k_a = 54 \quad h_b > h_r,$
$k_a = 54 - 0.8h_b \quad R \ge 500$ m, $h_b \le h_r,$
$k_a = 54 - 1.6\Delta h_b R \quad R < 500$ m, $h_b \le h_r.$

Note: Both L_{bsh} and k_a increase the path loss with lower base station antenna heights.

$$k_d = 18 \quad h_b < h_r,$$

$$k_d = 18 - \frac{15\Delta h_b}{\Delta h_m} \quad h_b \ge h_r,$$

$$k_f = 4 + 0.7\left(\frac{f_c}{925} - 1\right) \text{ for midsize city and suburban area with moderate tree density,}$$

$$k_f = 4 + 1.5\left(\frac{f_c}{925} - 1\right) \text{ for metropolitan center.}$$

The range of parameters for which Walfisch-Ikegami model is valid follows:

$$800 \le f_c \le 2,000 \text{ MHz}$$

$$4 \le h_b \le 50 \text{ m}$$

$$1 \le h_m \le 3 \text{ m}$$

$$0.02 \le R \le 5 \text{ km}$$

The following default values can be used for the model:

b = 20 – 50 m,
W = $b/2$,
ϕ = 90 degree,
Roof = 3 m for pitched roof and 0 m for flat roof,
h_r = 3 (number of floors) + Roof.

Using the following data, a comparison of the path loss from the Hata and Walfisch-Ikegami models is made in table 9.1.

f_c = 880 MHz,
h_m = 1.5 m,
h_b = 30 m,
Roof = 0 m,
h_r = 30 m,
ϕ = 90 degree,
b = 30 m,
W = 15 m.

Table 9.1 A Comparison of Path Loss from Hata and Walfisch-Ikegami Model

	Path Loss, dB	
Distance, km	Hata's Model	Walfisch–Ikegami Model
1	126.16	139.45
2	136.77	150.89
3	142.97	157.58
4	147.37	162.33
5	150.79	166.01

The path losses predicted by Hata's model are 13–16 dB lower than those predicted by the Walfisch–Ikegami model. Hata's model ignores effects from street width, street diffraction, and scatter losses, which the Walfisch–Ikegami model includes.

Correction Factor for Attentuation Due to Trees. Weissberger [13] has developed a modified exponential delay model that can be used where a radio path is blocked by dense, dry, in-leaf trees found in temperate climates. The additional path loss can be calculated from the following expression:

$$L_f = 1.33(f_c)^{0.284}(d_f)^{0.588} \text{ dB} \quad \text{for } 14 \le d_f \text{ m} \tag{9.13}$$

$$= 0.45(f_c)^{0.284} d_f \text{ dB} \quad \text{for } 0 \le d_f \le 14 \text{ m}$$

where L_f = loss in dB,
f_c = frequency in GHz,
d_f = tree height in meters.

The difference in path loss for trees with and without leaves has been found to be about 3–5 dB. For a frequency of 900 MHz, equation (9.13) is reduced to

$$
\begin{aligned}
L_f &= 1.291(d_f)^{0.588}\ \text{dB} \qquad \text{for } 14 \le d_f \le 400\ \text{m} \qquad (9.14)\\
&= 0.437 d_f\ \text{dB} \qquad \text{for } 0 \le d_f \le 14\ \text{m}
\end{aligned}
$$

9.3.2 Models for Indoor Environment

Experimental studies have indicated that a portable receiver moving in a building experiences Rayleigh fading for obstructed propagation paths and Ricean fading for line of sight (LOS) paths, regardless of the type of building. *Rayleigh fading* is short-term fading resulting from signals traveling separate paths (multipath) that partially cancel each other. A LOS path is clear of building obstructions; in other words, there are no reflections of the signal. *Ricean fading* results from the combination of a strong LOS path and a ground path plus numerous weak reflected paths. A TIA IS-95A mobile station, however, is capable of discerning signals that travel different paths because a RAKE receiver is incorporated. In fact, TIA IS-95A does not address equalization of delay spread on the radio link, and thus the mobile station's receiver does not have an equalizer.

Quantification of propagation between floors is important for in-building wireless system of multifloor buildings that need to share frequencies within the building. Frequencies are reused on different floors to avoid co-channel interference. The type of building material, the aspect ratio of the building sides, and the types of windows have been shown to have an impact on the RF attenuation between floors. Measurements have indicated that loss between floors does not increase linearly in dB with increasing separation distance. The greatest floor attenuation in dB occurs when the receiver and transmitter are separated by a single floor. The overall path loss increases at a smaller rate as the numbers of floors increase. Typical values of attentuation between floors is 15 dB for one floor of separation and an additional 6–10 dB per floor separation up to four floors of separation. For five or more floors of separation, the path loss will increase by only a few dB for each additional floor (see table 9.2).

Table 9.2 Mean Path Loss Exponents and Standard Deviations

Type	n	σ, dB
All building		
All locations	3.14	16.3
Same floor	2.76	12.9
Through 1 floor	4.19	5.1
Through 2 floors	5.04	6.5
Office building 1		
Entire building	3.54	12.8
Same floor	3.27	11.2
West wing 5th floor	2.68	8.1
Central wing 5th floor	4.01	4.3
West wing 4th floor	3.18	4.4
Grocery store	1.81	5.2
Retail store	2.18	8.7
Office building 2		
Entire building	4.33	13.3
Same floor	3.25	5.2

The signal strength received inside of a building due to an external transmitter is important for wireless systems that share frequencies with neighboring buildings or with an outdoor system. Experimental studies have shown that the signal strength received inside a building increases with height. At lower floors of a building, the urban cluster induces greater attenuation and reduces the level of penetration. At higher floors, an LOS path may exist, thus causing a stronger incident signal at the exterior wall of the building. RF penetration is found to be a function of frequency as well as height within a building. Penetration loss decreases with increasing frequency. Measurements made in front of a window showed 6 dB less penetration loss on coverage than those measurements made in parts of the building without window. Experimental studies also showed that building penetration loss decreased at a rate of about 2 dB per floor from ground level up to the 10th floor and then began to increase around the 10th floor. The increase in penetration loss at the higher floors was attributed to shadowing effects of adjacent buildings.

The mean path loss is a function of distance to the nth power [8]:

$$L_{50}(R) = L(R_0) + 10 \times n \log\left(\frac{R}{R_0}\right) \text{ dB} \tag{9.15}$$

where $L_{50}(R)$ = mean path loss (dB),
$L(R_0)$ = path loss from transmitter to reference distance R_0 (dB),
n = mean path loss exponent,
R = distance from the transmitter (m),
R_0 = reference distance from the transmitter (m).

We choose R_0 equal to 1 m and assume $L(R_0)$ due to free-space propagation from the transmitter to be a 1-m reference distance. Next, we assume that the antenna gain equals the system cable losses[3] and get a path loss $L(R_0)$, of 31.7 dB at 914 MHz over a 1-m free-space path.

The path loss was found to be lognormally distributed about equation (9.15). The mean path loss exponent n and standard deviation σ are the parameters that depend on building type, building wing, and number of floors between the transmitter and receiver. The path loss at a transmitter-receiver (T-R) separation of R meters can be given as

$$L(R) = L_{50}(R) + X_\sigma \text{ dB} \tag{9.16}$$

where $L(R)$ = path loss at a T-R separation distance R meters,
X_σ = zero mean lognormally distributed random variable with standard deviation σ dB.

Table 9.2 gives a summary of the mean path loss exponents and standard deviation about the mean for different environments [8].

In the multifloor environment, equation (9.15) can be modified to emphasize the mean path loss exponent as a function of the number of floors between transmitter and receiver. The value of n (multifloor) is given in table 9.2.

$$L_{50}(R) = L(R_0) + 10 \times n(multifloor) \log\left(\frac{R}{R_0}\right) \tag{9.17}$$

Another path loss prediction model suggested in [8] uses the floor attenuation factor (FAF). A constant floor attenuation factor (in dB), which is a function of the number of floors and building type, was

3. This is obviously not always true.

included in the mean path loss predicted by a path loss model that uses the *same floor* path loss exponent for the particular building type.

$$L_{50}(R) = L(R_0) + 10 \times n(same - floor) \log \left(\frac{R}{R_0}\right) + \text{FAF dB} \quad (9.18)$$

where R is in meters and $L(R_0) = 31.7$ dB at 914 MHz.

Table 9.3 provides the floor attentuation factors and standard deviation (in dB) of the difference between the measured and predicted path loss. Values for the floor attentuation factor in table 9.3 are an average (in dB) of the difference between the path loss observed at multifloor locations and the mean path loss predicted by the simple R^n model [equation (9.15)], where n is the *same floor* exponent listed in table 9.2 for the particular building structure and R is the shortest distance measured in three dimensions between the transmitter and the receiver.

Table 9.3 Average Floor Attenuation Factors

	FAF, dB	σ
Office building 1		
Through 1 floor	12.9	7.0
Through 2 floors	18.7	2.8
Through 3 floors	24.4	1.7
Through 4 floors	27.0	1.5
Office building 2		
Through 1 floor	16.2	2.9
Through 2 floors	27.5	5.4
Through 3 floors	31.6	7.2

Soft Partition and Concrete Wall Attenuation Factor Model. The path loss effects of *soft partitions* and *concrete walls* (in dB) between the transmitter and receiver for the same floor was modeled in [5] and has been given as

$$L_{50}(R) = 20 \log\left(\frac{4\pi R}{\lambda}\right) + p \times AF(soft - partition) + q \times AF(concrete - wall) \quad (9.19)$$

where: p = number of soft partitions between the transmitter and the receiver,
q = number of concrete walls between the transmitter and the receiver,
λ = wavelength (m),
AF = 1.39 dB for each soft partition,
AF = 2.38 for each concrete wall.

Example 9.1

Use the two models [equations (9.17) and (9.18)] to predict the mean path loss at a distance R = 30 m through three floors of an office building. Assume the mean path loss exponent for *same floor* measurements in the building is n = 3.27, the mean path loss exponent for three-floor measurements is n = 5.22, and the average FAF is 24.4 dB.

From equation (9.17),

$$L_{50}(30) = 31.7 + 10 \times 5.22 \log\left(\frac{30}{1}\right) = 108.8 \text{ dB}$$

From equation (9.18),

$$L_{50}(30) = 31.7 + 10 \times 3.27 \log\left(\frac{30}{1}\right) + 24.4 = 104.4 \text{ dB}$$

The results obtained by the two models are fairly close.

9.4 Link Budgets

The link budgets are intended to provide necessary calculations for the ratio of the received bit energy (E_b) to the thermal noise (N_0) plus interference density (I_0) based on the transmit power, transmit and receive antenna gains, receiver noise figure, channel capacity factor, and propagation as well as interference environment.

Link budgets are required for the forward and reverse traffic channels, pilot channel, paging channel, and sync channel.

9.4.1 Forward Direction

To calculate the effective signal-to-interference ratio for the forward pilot, sync, paging, and traffic channels, we need to calculate the received signal power and received interference on each channel. The following equations enable us to complete the calculations.

1. Traffic Channel Effective Radiated Power:

$$p_t = \frac{P_t}{N_t C_f} \tag{9.20}$$

or

$$p_t = P_t - 10 \log N_t - 10 \log C_f \qquad \text{(dBm)} \tag{9.21}$$

where p_t = the effective radiated power (ERP) of the traffic channel (dBm),
P_t = the ERP of all traffic channels from the transmit antenna of cell site (dBm),
N_t = the number of traffic channels supported by the sector or cell,
C_f = the channel voice activity factor (typical value is 0.4–0.6)

2. Power per User:

$$p_u = p_t - G_t - L_c \qquad \text{(dBm)} \tag{9.22}$$

where p_u = the traffic channel power per user (dBm),
G_t = the gain of the cell transmit antenna (dB), and
L_c = the transmit filter and cable loss between the output of the linear amplifier circuit (LAC) and the input of transmit antenna (dB).

3. Total Base Station ERP:

$$P_c = 10 \log[10^{0.1 p_t} + 10^{0.1 p_s} + 10^{0.1 p_p} + 10^{0.1 p_{pg}}] \qquad \text{(dBm)} \tag{9.23}$$

where P_c = the total base station ERP (dBm),
p_s = the ERP of the sync channel (dBm),
p_p = the ERP of the pilot channel (dBm),
p_{pg} = the ERP of the paging channel (dBm).

4. Base Station Power Amplifier:

$$P_a = P_c - G_t - L_c \tag{9.24}$$

where P_a = the total power of all traffic channels, pilot, paging, and sync channels at the output of the amplifier.

5. Total Mobile Received Power:

$$p_m = P_c + L_p + A_l + G_m + L_m \qquad \text{(dBm)} \tag{9.25}$$

where p_m = the total mobile received power (dBm),

L_p = the average propagation path loss between the base station and the mobile (dB),
A_l = the allowance for lognormal shadow loss due to local terrain for a given coverage probability (dB),
G_m = the (receive) gain of the mobile antenna (dB),
L_m = the mobile receiver cable and connector losses (dB).

6. Received Traffic Channel Power:

$$p_{tr} = p_t + L_p + A_l + G_m + L_m \qquad \text{(dBm)} \tag{9.26}$$

where p_{tr} = the received power of traffic channel by the mobile from the serving base station.

7. Received Pilot Power:

$$p_{pr} = p_p + L_p + A_l + G_m + L_m \qquad \text{(dBm)} \tag{9.27}$$

where p_{pr} = the received power of the pilot channel by the mobile from the serving base station.

8. Received Paging Channel Power:

$$p_{pgr} = p_{pg} + L_p + A_l + G_m + L_m \qquad \text{(dBm)} \tag{9.28}$$

where p_{pgr} = the received power of the paging channel by the mobile from the serving base station.

9. Received Sync Channel Power:

$$p_{sr} = p_s + L_p + A_l + G_m + L_m \qquad \text{(dBm)} \tag{9.29}$$

where p_{sr} = the received power of the sync channel by the mobile from the serving base station.

10. Interference from Other Users (same Base Station) on the Traffic Channel:

$$I_{ut} = 10 \log[10^{0.1p_m} - 10^{0.1p_{tr}}] - 10 \log B_w \tag{9.30}$$

where I_{ut} = the density of interference from other users on the traffic channel (dBm/Hz),
B_W = the bandwidth (Hz).

11. Interference from Other Base Stations on the Traffic Channel:

$$I_{ct} = I_{ut} + 10 \log\left[\frac{1}{f_r} - 1\right] \tag{9.31}$$

where I_{ct} = the density of interference from other base stations on the traffic channel (dBm/Hz),
f_r = frequency reuse factor (typical value is 0.65).

12. Interference Density for the Traffic Channel:

$$I_t = 10 \log[10^{0.1 I_{ut}} + 10^{0.1 I_{ct}}] \quad (9.32)$$

where I_t = the density of interference on the traffic channel (dBm/Hz).

13. Interference from Other Users (Same Base Station) to the Pilot Channel:

$$I_{up} = p_m - 10 \log B_w \quad (9.33)$$

where I_{up} = the density of interference from other users on the pilot channel (dBm/Hz).

14. Interference from Other Base Stations on the Pilot Channel:

$$I_{cp} = I_{up} + 10 \log\left[\frac{1}{f_r} - 1\right] \quad (9.34)$$

where I_{cp} = the density of interference from other base stations on the pilot channel (dBm/Hz).

15. Interference Density for the Pilot Channel:

$$I_p = 10 \log[10^{0.1 I_{up}} + 10^{0.1 I_{cp}}] \quad (9.35)$$

where I_p = the density of interference for the pilot channel (dBm/Hz).

16. Interference from Other Users (Same Base Station) to the Paging Channel:

$$I_{upg} = 10 \log[10^{0.1 p_m} - 10^{0.1 p_{pgr}}] - 10 \log B_w \quad (9.36)$$

where I_{upg} = the density of interference from other users on the paging channel (dBm/Hz).

17. Interference from other Base Stations on the Paging Channel:

$$I_{cpg} = I_{upg} + 10 \log\left[\frac{1}{f_r} - 1\right] \quad (9.37)$$

where I_{cpg} = the density of interference from other base stations on the paging channel (dBm/Hz)

18. Interference Density for the Paging Channel:

$$I_{pg} = 10 \log[10^{0.1 I_{upg}} + 10^{0.1 I_{cpg}}] \tag{9.38}$$

where I_{pg} =density of interference for the paging channel (dBm/Hz)

19. Interference from Other Users (Same Base Station) on the Sync Channel:

$$I_{us} = 10 \log[10^{0.1 p_m} - 10^{0.1 p_{sr}}] - 10 \log B_w \tag{9.39}$$

where I_{us} = the density of interference from other users on the sync channel (dBm/Hz).

20. Interference from Other Base Stations on the Sync Channel:

$$I_{cs} = I_{us} + 10 \log\left[\frac{1}{f_r} - 1\right] \tag{9.40}$$

where I_{cs} = the density of interference from other base stations on the sync channel (dBm/Hz).

21. Interference Density for the Sync Channel:

$$I_s = 10 \log[10^{0.1 I_{us}} + 10^{0.1 I_{cs}}] \tag{9.41}$$

where I_s = the density of interference on the sync channel (dBm/Hz).

22. Thermal Noise:

$$N_0 = 10 \log(290 \times 1.38 \times 10^{-23}) + N_f + 30 \tag{9.42}$$

where N_0 = thermal noise density (dBm/Hz),
N_f = noise figure of mobile receiver (dB).

23. Traffic Channel Signal-to-Noise Plus Interference Ratio:

$$\frac{E_b}{N_0 + I_t} = p_{tr} - 10 \log b_{rt} - 10 \log[10^{0.1 I_t} + 10^{0.1 N_0}] \tag{9.43}$$

where b_{rt} = the bit rate of the traffic channel (bps).

24. Pilot Channel Signal-to-Noise Plus Interference Ratio:

$$\frac{E_b}{N_0 + I_p} = p_{pr} - 10 \log B_w - 10 \log[10^{0.1I_p} + 10^{0.1N_0}] \tag{9.44}$$

25. Paging Channel Signal-to-Noise Plus Interference Ratio:

$$\frac{E_b}{N_0 + I_{pg}} = p_{pgr} - 10 \log b_{rpg} - 10 \log[10^{0.1I_{pg}} + 10^{0.1N_0}] \tag{9.45}$$

where b_{rpg} = the bit rate of the paging channel (bps).

26. Sync Channel Signal-to-Noise Plus Interference Ratio:

$$\frac{E_b}{N_0 + I_s} = p_{sr} - 10 \log b_{rs} - 10 \log[10^{0.1I_s} + 10^{0.1N_0}] \tag{9.46}$$

where b_{rs} = the bit rate of the sync channel (bps).

9.4.2 Reverse Direction

The calculations on the reverse channel are similar to calculations on the forward channels; we use the following equations.

1. Mobile Power Amplifier:

$$P_{ma} = P_{me} - L_m - G_m \tag{9.47}$$

where P_{ma} = the power output of the mobile power amplifier (dBm),[4]
P_{me} = the ERP from the transmit antenna of the mobile (dBm),
L_m = the transmit filter and cable loss between the output of the power amplifier and input of the transmit antenna of the mobile (dB),
G_m = the gain of the mobile transmit antenna (dB).

2. Base Station Received Power per User:

$$P_{cu} = P_{me} + L_p + A_l + G_t + L_t \tag{9.48}$$

where P_{cu} = the received power of a traffic channel from a mobile by the serving base station (dBm),
L_p = the mean propagation path loss between the mobile and the base station (dB),

4. dBm is referenced to 1 milliwatt.

A_l = the lognormal shadow/fade allowance due to local terrain for a given coverage probability (dB),
G_t = the (receive) gain of the base station antenna,
L_t = the base station receiver cable and connector losses (dB),

3. Interference Density of Other Mobiles in the Serving Base Station:

$$I_{utr} = P_{cu} + 10 \log(N_t - 1) + 10 \log C_a - 10 \log(B_w) \tag{9.49}$$

where I_{utr} = the interference density from other mobiles in the serving base station (dBm/Hz),
C_a = the channel voice activity factor (typical value is 0.4–0.6),
N_t = the number of traffic channels supported per sector/base station,
B_W = the bandwidth (Hz).

4. Interference Density from the Mobiles in Other Base Stations:

$$I_{ctr} = I_{utr} + 10 \log\left(\frac{1}{f_r} - 1\right) \tag{9.50}$$

where I_{ctr} = the density of interference from the mobiles in other base stations (dBm/Hz),
f_r = frequency reuse factor (typical value = 0.6).

5. Interference Density from Other Mobiles in the Serving Base Station and Other Base Stations:

$$I_{tr} = 10 \log[10^{0.1 I_{utr}} + 10^{0.1 I_{ctr}}] \tag{9.51}$$

where I_{tr} = the density of interference density from other mobiles in the serving base station and from other base stations (dBm/Hz).

6. Thermal Noise Density:

$$N_0 = 10 \log(290 \times 1.38 \times 10^{-23}) + N_f + 30 \tag{9.52}$$

where N_0 = the thermal noise density at the reference thermal noise temperature of 290 K,
N_f = the noise figure of the base station receiver (dB).

7. Reverse Traffic Channel Signal-to-Noise Plus Interference Ratio:

$$\frac{E_b}{N_0 + I_{tr}} = P_{cu} - 10 \log b_{rr} - 10 \log[10^{0.1 I_{tr}} + 10^{0.1 N_0}] \tag{9.53}$$

where b_{rr} = the bit rate on the reverse traffic channel (bps).

Example 9.2

Use the following data and perform necessary calculations to develop link budgets in the forward and reverse direction:

Total traffic channels ERP	=	57 dBm
Mobile ERP	=	20 dBm
Numbers of users	=	20
Traffic channel activity factor	=	0.6
Pilot channel ERP	=	51.5 dBm
Paging channel ERP	=	46.94 dBm
Sync channel ERP	=	41.5 dBm
Transmit filter and cable losses	=	–2.5 dB
Mobile receive cable and body losses	=	–3 dB
Cell transmit antenna gain	=	14 dB
Mobile antenna gain	=	0 dB
Mean propagation losses	=	–146 dB
Lognormal shadow/fade allowance	=	–6.2 dB
Base station noise figure	=	5 dB
Mobile noise figure	=	8 dB
Forward traffic channel bit rate	=	9600 bps
Reverse traffic channel bit rate	=	9600 bps
Paging channel bit rate	=	9600 bps
Sync channel bit rate	=	9600 bps
Reverse traffic channel bit rate	=	9600 bps

Forward Direction:

$$p_t = 57 - 10 \log 20 - 10 \log 0.6 = 57 - 13.01 + 2.22 = 46.21 \text{ dBm}$$

$$P_c = 10 \log[10^{5.7} + 10^{5.15} + 10^{4.69} + 10^{4.15}] = 58.49 \text{ dBm}$$

$$p_u = 46.21 - 14 + 2.5 = 34.71 \text{ dBm}$$

$$P_a = 58.49 - 14 + 2.5 = 46.99 \text{ dBm}$$

$$p_m = 58.49 - 146 - 6.2 - 0 - 3 = -96.71 \text{ dBm}$$

$$p_{tr} = 46.21 - 146 - 6.2 - 0 - 3 = -108.99 \text{ dBm}$$

$$p_{pr} = 51.5 - 146 - 6.2 - 0 - 3 = -103.70 \text{ dBm}$$

$$p_{pgr} = 46.94 - 146 - 6.2 - 0 - 3 = -108.26 \text{ dBm}$$

$$p_{sr} = 41.5 - 146 - 6.2 - 0 - 3 = -113.70 \text{ dBm}$$

$$I_{ut} = 10 \log[10^{-9.671} - 10^{-10.899}] - 10 \log(1.2288 \times 10^6) = -157.87 \text{ dBm/Hz}$$

$$I_{ct} = -157.87 + 10 \log\left[\frac{1}{0.65} - 1\right] = -160.56 \text{ dBm/Hz}$$

$$I_t = 10 \log[10^{-15.787} + 10^{-16.056}] = -156 \text{ dBm/Hz}$$

$$I_{up} = -96.71 - 60.895 = -157.605 \text{ dBm/Hz}$$

$$I_{cp} = -157.605 + 10 \log\left[\frac{1}{0.65} - 1\right] = -160.295 \text{ dBm/Hz}$$

$$I_p = 10 \log[10^{-15.7605} + 10^{-16.0295}] = 155.73 \text{ dBm/Hz}$$

$$I_{upg} = 10 \log[10^{-9.671} - 10^{-10.826}] - 10 \log(1.2288 \times 10^6) = -157.92 \text{ dBm/Hz}$$

$$I_{cpg} = -157.92 + 10 \log\left[\frac{1}{0.65} - 1\right] = -160.61 \text{ dBm/Hz}$$

$$I_{pg} = 10 \log[10^{-15.792} + 10^{-16.061}] = -156.05 \text{ dBm/Hz}$$

$$I_{us} = 10 \log[10^{-9.671} - 10^{-11.37}] - 10 \log(1.2288 \times 10^6) = -157.693 \text{ dBm/Hz}$$

$$I_{cs} = -157.693 + 10 \log\left[\frac{1}{0.65} - 1\right] = -160.382 \text{ dBm/Hz}$$

$$I_s = 10 \log[10^{-15.7693} + 10^{-16.0382}] = -155.82 \text{ dBm/Hz}$$

$$N_0 = 10 \log(290 \times 1.38 \times 10^{-23}) + 8 + 30 = -165.98 \text{ dBm/Hz}$$

Traffic Channel:

$$\frac{E_b}{N_0 + I_t} = -108.99 - 10 \log(9600) - 10 \log[10^{-15.6} + 10^{-16.598}] = 6.764 \text{ dB}$$ [5]

5. Typical E_b/I_0 value for the traffic channel is between 6 and 7 dB.

Pilot Channel:

$$\frac{E_b}{N_0 + I_p} = -103.7 - 60.685 - 10\log[10^{-15.573} + 10^{-16.598}] = -9.257 \text{ dB}^{6}$$

Paging Channel:

$$\frac{E_b}{N_0 + I_{pg}} = -108.26 - 39.83 - 10\log[10^{-15.605} + 10^{-16.598}] = 7.54 \text{ dB}$$

Sync Channel:

$$\frac{E_b}{N_0 + I_s} = -113.7 - 30.792 - 10\log[10^{-15.582} + 10^{-16.598}] = 10.93 \text{ dB}$$

Reverse Direction:

$$P_{ma} = 20 - (-3) - 0 = 23 \text{ dBm}$$

$$P_{cu} = 20 - 146 - 6.2 + 14 - 2.5 = -120.7 \text{ dBm}$$

$$I_{utr} = -120.7 + 10\log(20-1) + 10\log 0.6 - 10\log(1.2288 \times 10^{6}) = -171.03 \text{ dBm/Hz}$$

$$I_{ctr} = -171.03 + 10\log\left[\frac{1}{0.6} - 1\right] = -172.79 \text{ dBm/Hz}$$

$$I_{tr} = 10\log[10^{-17.103} + 10^{-17.279}] = -168.8 \text{ dBm/Hz}$$

$$N_0 = 10\log(290 \times 1.38 \times 10^{-23}) + 5 + 30 = -168.98 \text{ dBm/Hz}$$

Traffic Channel:

$$\frac{E_b}{N_0 + I_{tr}} = -120.7 - 10\log 9600 - 10\log[10^{-16.88} + 10^{-16.898}] = 5.35 \text{ dB}$$

9.5 Dual-Mode CDMA Mobiles

The nominal CDMA channel requires 41 contiguous analog channels with a 9-channel spacing as a guard between the edge of a CDMA channel and the adjacent analog channels. This implies that 59 contiguous analog channels should be removed from the service to introduce the first

6. The pilot channel does not carry any information and is used only for pilot acquisition. Thus, the corresponding E_b/I_0 for pilot may be less than the Shannon's limit discussed in chapter 2.

CDMA channel, 100 analog channels for the first and second CDMA channel, etc. In an analog system with a reuse factor of 7, three analog channels per sector, per base station, will be removed to add first CDMA channel (i.e., $(59/7)/3 \approx 3$) and an additional 2 analog channels per sector for each additional CDMA channel (i.e., $(42/7)/3 = 2$).

We consider an example where the base station is equipped with 78 channel elements. Four channel elements on each sector are dedicated to the pilot channel, the sync channel, and two paging/access channels, leaving 22 channel elements per sector remaining for traffic channels. We assume that 22 channel elements are permanently assigned to each sector. The real system pools all 66 channel elements so that any channel element can be assigned to any sector; this method is referred to as dynamic channel assignment. *Dynamic channel assignment* improves the base station capacity as it reduces the probability of blocking on a sector since an idle channel element can be used in any sector. Blocking occurs when the interference from an additional base station will reduce the voice quality below the acceptable limit. Blocking is determined by the channel capacity and not by the amount of hardware used. Using the method of permanently assigning channel elements to a sector gives the lower bound on the total traffic carried by the base station. The traffic carried by the same channel elements will be higher in the real system.

We consider the case where a three-sector analog base station (BS) is equipped with 57 channels that are assigned an $N = 7$ reuse pattern. We remove 3 analog channels to introduce one CDMA channel to serve 22 users. We assume no overflow traffic from CDMA to analog channels and 2 percent probability of blocking and model the traffic using Erlang B-statistic. We also assume that each sector can serve 22 simultaneous calls with the same voice quality as an analog channel. Table 9.4 provides the calculated capacity of the analog base station and the combined CDMA and analog base station. It may be noted that the traffic capacity of the combined base station is double the capacity of the analog base station.

Table 9.4 Comparison of BS capacity with Analog and Combined CDMA and Analog Channels

Analog Voice Channels	CDMA Traffic Channels	Analog Traffic per sector, Erlangs	Total Analog Traffic, Erlangs	CDMA Traffic per sector, Erlangs	Total CDMA Traffic, Erlangs	Total BS Traffic, Erlangs
19	0	12.34	37.0	—	—	37.0
16	22	9.83	29.5	14.9	44.7	74.2

Example 9.3

Estimate the base station capacity as a function of CDMA user penetration for an overlay system with 2 percent blocking for both analog and digital subscribers. Assume $N = 7$ as the reuse pattern for the analog channels and $N = 1$ for CDMA channels. The average traffic per subscriber is 0.02 Erlangs. See table 9.5.

Introducing CDMA in an analog system results in reducing the analog base station capacity by 20 percent but doubling the total capacity of the base station. This means that at least 20 percent of the offered traffic load must come from dual-mode mobiles before activating CDMA. To realize 100 percent capacity increase in the base station, about 60 percent of the offered traffic load in the base station must be from dual-mode mobiles.

9.6 The Transition from an Analog System to a Digital System

The transition from the analog system to the CDMA system as offered by the cellular system equipment vendors and envisioned by the cellular system operators generally falls into three basic modes:

- An overlay design of two separate, independent systems: one is analog and the other is CDMA.
- A completely integrated system where CDMA and analog service is offered everywhere.
- A partially integrated system in which CDMA and analog coverage is provided in part of the overall service area, and only analog coverage exists in the remaining service area.

9.6.1 Overlay Design

In the overlay scenario, a new CDMA system is superimposed over the existing analog system. The CDMA overlay could require a one-to-one digital base station for each analog base station, but it may use a smaller number of larger coverage area cells than the existing analog system to reduce the cost of base stations. Figure 9.2 shows a CDMA base station coverage area that is three times larger than the analog base station coverage area. Thus, it requires one third as many base stations to provide CDMA coverage over the service area. Such an overlay design could be operated as two separate independent systems that are operated by different vendors from separate mobile switching centers or as a single system in which a single MSC controls both types of base stations. Handoffs between the CDMA system and analog system are possible, and handoffs from analog system to CDMA system are not

Table 9.5 Capacity of a cell versus CDMA user Penetration

Analog Channels per Base Station, N_a	CDMA Channels per Base Station, N_c	CDMA Users per Base Station, CE[a]	Percent CDMA Traffic	Analog Mobile, Erlangs	CDMA Mobile, Erlangs	Total Erlangs	Analog Subscribers	CDMA Subscribers	Total Subscribers
57	0	0	0	$12.34 \times 3 =$ 37.0	0	37.0	1850	0	1850
48	1	66	60.24	$9.83 \times 3 =$ 29.5	$14.9 \times 3 =$ 44.7	74.2	1475	2235	3710
42	2	132	80.91	$8.2 \times 3 =$ 24.6	$34.68 \times 3 =$ 104.0	128.6	1230	5200	6430
36	3	198	89.3	$6.62 \times 3 =$ 19.9	$55.33 \times 3 =$ 166	185.9	995	8300	9295
30	4	264	93.73	$5.084 \times 3 =$ 15.3	$76.38 \times 3 =$ 229.1	244.4	765	11,455	12,220
24	5	330	96.43	$3.627 \times 3 =$ 10.9	$97.69 \times 3 =$ 293.1	304.0	545	14,655	15,200
18	6	396	98.13	$2.277 \times 3 =$ 6.8	$119.2 \times 3 =$ 357.6	364.4	340	17,880	18,220
12	7	462	99.22	$1.092 \times 3 =$ 3.3	$140.7 \times 3 =$ 422.2	425.5	165	21,110	21,275
6	8	528	99.86	$0.223 \times 3 =$ 0.7	$162.4 \times 3 =$ 487.3	488	35	24,365	24,400
0	9	594	100	0	$184.2 \times 3 =$ 552.5	552.5	0	27,625	27,625

a. CE = channel element.

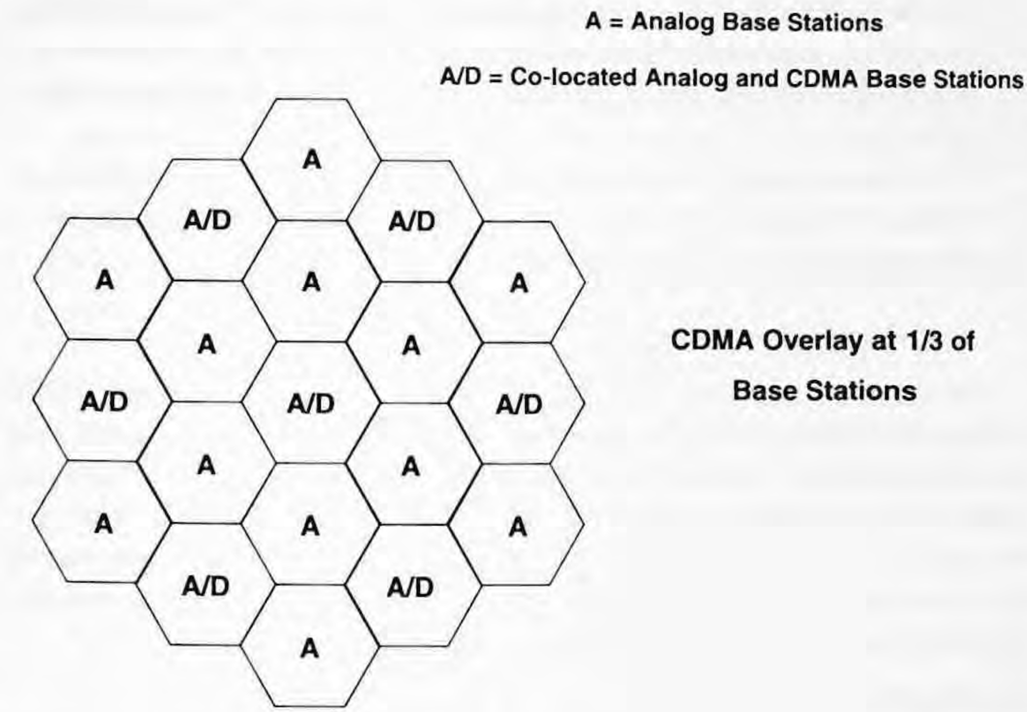

Figure 9.2 Overlay design.

allowed by the IS-95 standard. The advantages and disadvantages of the full overlay design follow:

Advantages:

- It allows independent analog and CDMA systems in the market.
- If desired, one can use a different vendor for the CDMA system, rather than the analog system vendor.
- The service provider will be able to advertise that CDMA (digital) service is available throughout their service area.
- Since a smaller number of base stations can be used, the investment in digital equipment should be lower than for designs that require digital equipment in a larger number of base stations.

Disadvantages:

- There is capacity loss due to segmentation of the cellular spectrum, since there will be fewer radio frequencies for the analog subscribers.
- The grade of service to analog subscribers, whose numbers may increase if analog terminal "dumping" occurs, will require investing in additional analog base station infrastructure.

- There is increased system operational complexity. The engineering, operation, administration, and maintenance (OA&M) of a two-system CDMA/analog overlay is much more complex than for a single system.
- The "analog-only" base stations may require additional RF filters to reduce the probability that a nearby CDMA mobile will overload the analog base station receiver.

9.6.2 Integrated Design

In this scenario (fig. 9.3), the system is designed to support both analog and CDMA customers everywhere in the service area. As with the other designs, this would be a transitional approach containing the capability for high CDMA capacity in the core and lower CDMA capacity in the noncore areas. Over time, the higher CDMA capacity area would expand to include more and more of the system. The advantages and disadvantages of this approach follow:

Advantages:

- The entire system would have complete digital coverage.
- This scenario avoids receiver overload and other radio problems.
- There is no cellular spectrum segmentation.

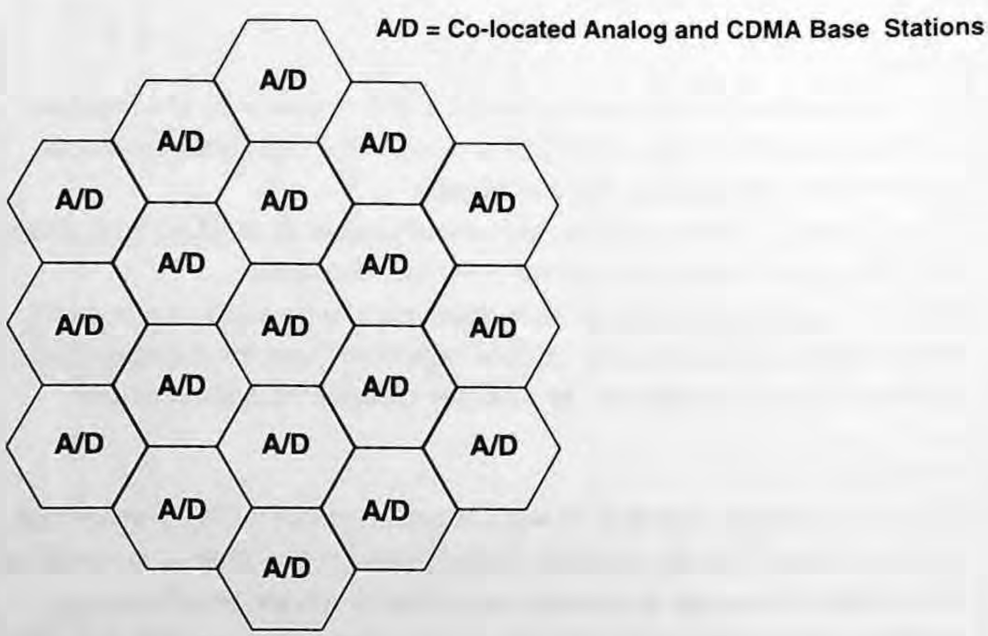

Figure 9.3 Integrated design.

- Full-spectrum efficiency is achieved through high reuse ($N = 1$) everywhere.
- Dual-mode terminals and handoffs between CDMA and analog channels are not required (except for roaming into analog coverage areas).
- OA&M is simplified from that of the overlay system.
- The system operator can advertise digital service everywhere in the service area.

Disadvantages:

- It requires digital equipment everywhere in the system. Although the investment costs can be reduced in areas that need only a low-capacity digital service, it is still a larger investment than for the partial digital system option.

9.6.3 Partial CDMA Coverage, Integrated System

In the partially integrated system, only a part of the system is converted to support analog and digital traffic (usually the core of the system most in need of traffic relief from the digital design). Surrounding the core area of the system, a transition or buffer zone is required to avoid interference between co-channel analog and digital channels. Co-channels are not allowed in the buffer zone. Beyond the buffer zone, base stations can be assigned analog channels that are co-channel with digital channels within the core. A dual-mode mobile that is assigned a digital channel in the core would be handed off to an analog channel as the mobile approaches the edge of the digital coverage area, and then it would be handed to a base station in the buffer zone. Mobiles assigned an analog channel in the analog-only base stations would not transition to a digital channel when inside the digital coverage area.

The transition zone can be gradually moved outward from the core, and the digital coverage can be expanded. The rate of expansion can be determined by the mix of terminals in the system, the need for capacity relief, the strategy for moving the customer base to digital terminals, and the economics of the system operator. Figure 9.4 shows an example of a simplified model with a uniform base station size. The buffer zone between the CDMA/analog base stations and analog-only base stations may require two tiers of base stations depending on the propagation and relative sizes of the actual base stations. If the base stations in the buffer zone are at full capacity, then adding CDMA will reduce the capacity of

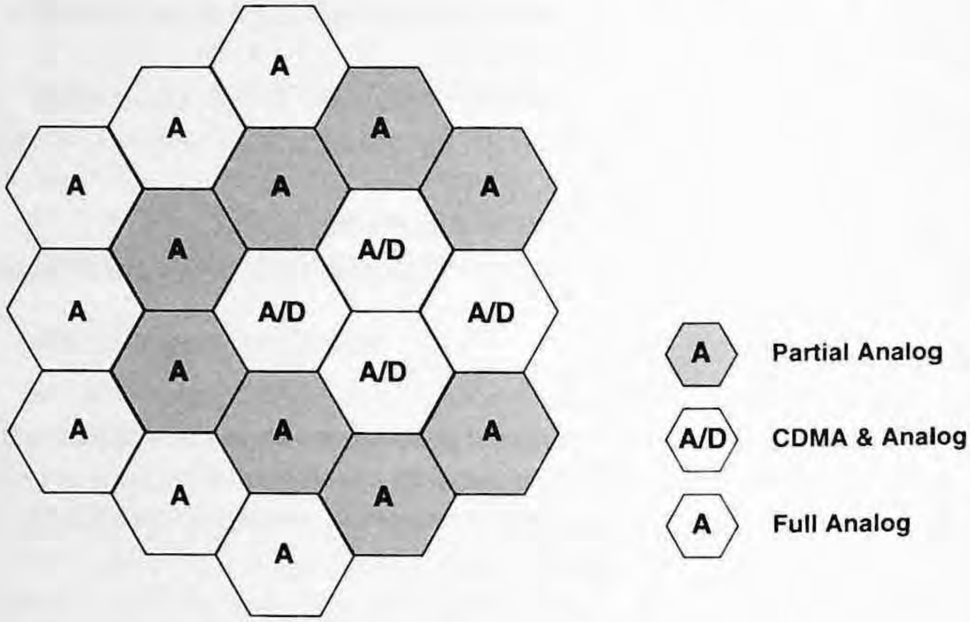

Figure 9.4 Partial CDMA coverage—integrated design.

the buffer zone base stations, since frequencies required by CDMA channels cannot be used in the buffer zone.

Figure 9.5 shows the case where a CDMA base station interferes with the analog mobile. The performance criterion for the analog mobiles is that the carrier signal from the analog base station is 17 dB greater than the total interference on the channel. If the analog system is designed for

$$\frac{S_{anal}}{I_{co-ch}} = 18 \text{ dB} \tag{9.54}$$

and if

$$\frac{S_{anal}}{I_{co-ch} + I_{CDMA}} \geq 17 \text{ dB} \tag{9.55}$$

provides adequate voice quality, then

$$\frac{S_{anal}}{I_{CDMA}} > 24 \text{ dB} \tag{9.56}$$

Table 9.6 lists the relative sizes of CDMA and analog base stations to achieve the criterion as a function of separation between the CDMA and analog center frequencies. With the recommended 9 guard channels,

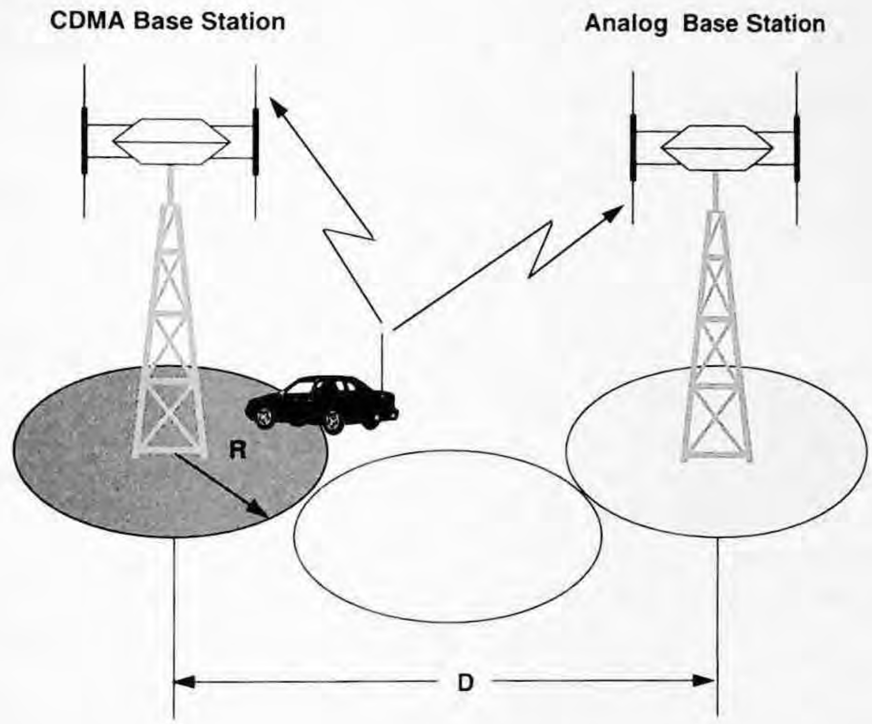

Figure 9.5 CDMA base station interference with analog mobile.

Table 9.6 Required *D/R* Ratios Versus Center Frequency Separation Between CDMA and Analog Center Frequency: CDMA BS Interferes with Analog MS

Center Frequency Separation (f_s), kHz	Required *D/R*
$f_s < 900$	≥ 2.90
$900 \leq f_s < 1980$	≥ 1.33
$f_s \geq 1980$	≥ 1.14

the CDMA base station has to be only slightly farther away than the radius of the analog base station to have adequate analog voice-quality performance. If the analog base station assigns frequencies within the CDMA channel, then the CDMA base station must be about three times the analog base station radius away, or a one base station buffer zone for equal size analog and CDMA base stations. (R is the radius of the analog base station, and D is the distance between analog and CDMA base stations.)

Figure 9.6 shows a case where the CDMA mobile interferes with the analog base station. The performance criterion for the analog base sta-

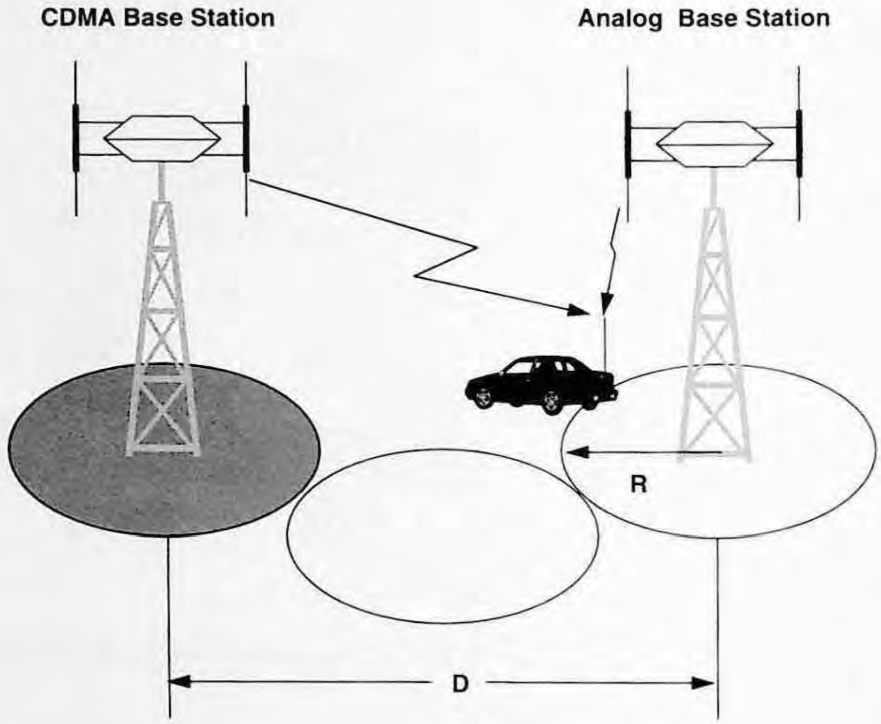

Figure 9.6 CDMA mobile interference with analog base station.

tion is that the received signal must be 17 dB greater than the total received noise plus interference. If the analog system is designed for

$$\frac{S_{anal}}{I_{co-ch}} = 18 \text{ dB} \tag{9.57}$$

and if

$$\frac{S_{anal}}{I_{co-ch} + I_{CDMA}} \geq 17 \text{ dB} \tag{9.58}$$

provides adequate voice quality, then

$$\frac{S_{anal}}{I_{CDMA}} > 24 \text{ dB} \tag{9.59}$$

Table 9.7 lists the required *D*/*R* ratios as function of the separation between the CDMA and analog center frequencies to achieve the criterion, where *D* = distance between the analog and CDMA base station and *R* = radius of the CDMA base station. With the recommended guard 9 channels, the CDMA mobile can be close to the analog base station before the interference becomes a problem. This situation is less critical than the previous case.

Table 9.7 Required *D/R* Ratio Versus Frequency Separation Between CDMA and Analog Center Frequencies: CDMA MS Interferes with Analog BS

Frequency Separation (f_s), kHz	Required *D/R*
$f_s < 900$	≥ 1.23
$900 \leq f_s < 1980$	≥ 1.05
$f_s \geq 1980$	≥ 1.02

Both rules (tables 9.6 and 9.7) must be satisfied when deploying CDMA base stations to prevent interference to the existing analog base stations. Meeting these criteria will ensure that the interference from the analog transmitters to the CDMA receivers is not a problem as well.

The following are the advantages and disadvantages of the partial integrated design:

Advantages:

- The CDMA capacity advantage can be placed where it is needed most (i.e., in the core system). Only investment for the core system will be needed.
- OA&M is simpler than in the overlay approach.
- The design avoids the receiver overload problem of the overlay design.

Disadvantages:

- The system operator cannot advertise "digital everywhere" service.
- Handoff is required between CDMA and analog coverage areas.
- IS-95 does not provide analog to CDMA handback, so a call initiated in the analog area will not provide any digital features available in the digital coverage area.
- Voice quality changes may be perceived during CDMA to analog handoffs.

Example 9.4

We consider a small city cellular system that is growing at the predicted growth rate for the next 7 years (see table 9.8). The startup system required 9 omnidirectional coverage base stations and has grown to 29 directional analog base stations to provide service for 36,000 busy hour call attempts (BHCA). Based on the predictions, this system must be expanded to provide capacity for 100,000 BHCA at the end of 7 years. The service provider has chosen to provide CDMA service over the complete coverage area by overlaying the coverage with 10 base stations. The service provider reduces the analog subscribers gradually as indicated in table 9.8. The traffic per subscriber during the busy hour is 0.02 Erlangs.

Table 9.8 Prediction for Analog and Digital Subscribers

End of Year	Analog Subscribers	CDMA Subscribers	Total Subscribers
0	36,000	0	36,000
1	29,000	13,000	42,000
2	24,000	22,000	46,000
3	16,000	39,000	55,000
4	10,000	52,000	62,000
5	4,000	66,000	70,000
6	0	82,000	82,000
7	0	100,000	100,000

Table 9.9 At the End of Year 0

No. of Base Stations		No. of RF Channels/Sector		Capacity, Erlangs/sector		
Analog	CDMA	Analog	CDMA	Analog	CDMA	BHCA
29	—	14	—	8.2	—	36,000

Year 0 (all Analog) (see fig. 9.7 and table 9.9):

- Total traffic during busy hour: $36{,}000 \times 0.02 = 720.0$ Erlangs
- Traffic per sector: $720.0 / (29 \times 3) = 8.276$ Erlangs
- Number of voice channels per sector to provide 2 percent blocking: 14

Year 1 (see fig. 9.8 and table 9.10):

- CDMA traffic: $13{,}000 \times 0.02 = 260$ Erlangs
- Analog traffic: $29{,}000 \times 0.02 = 580.0$ Erlangs

To provide full CDMA coverage, a CDMA minicell is added to 10 base stations.

- CDMA traffic per sector: $260.0 / (10 \times 3) = 8.67$ Erlangs

One CDMA channel provides 14.9 Erlangs per sector with 2 percent system blocking to serve the dual-mode mobiles, so the service provider decides to eliminate all analog channels in the expanded spectrum and uses the spectrum for one CDMA channel in the minicell equipment. This reduces the analog capacity by 3 channels per sector in all 29 base stations.

- Analog traffic per sector: $(29{,}000 \times 0.02) / (29 \times 3) = 6.67$ Erlangs
- Number of analog channel per sector: $14 - 3 = 11$

These 11 channels will carry 6.83 Erlangs of traffic at 4 percent blocking for the analog subscribers. The single CDMA channel will carry all the offered

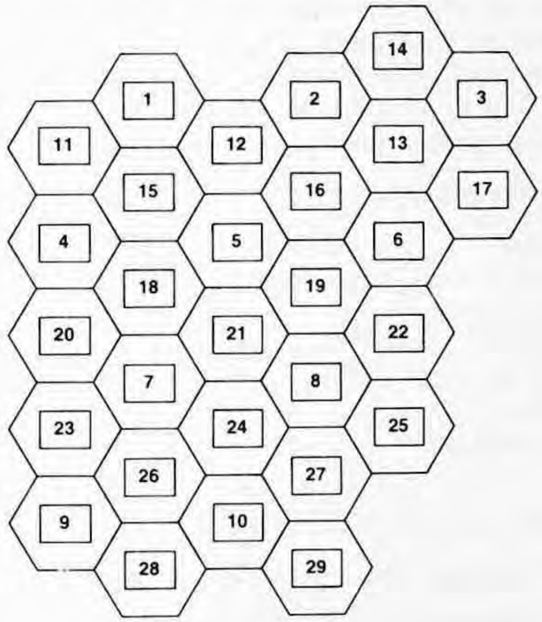

Figure 9.7 Configuration at year 0.

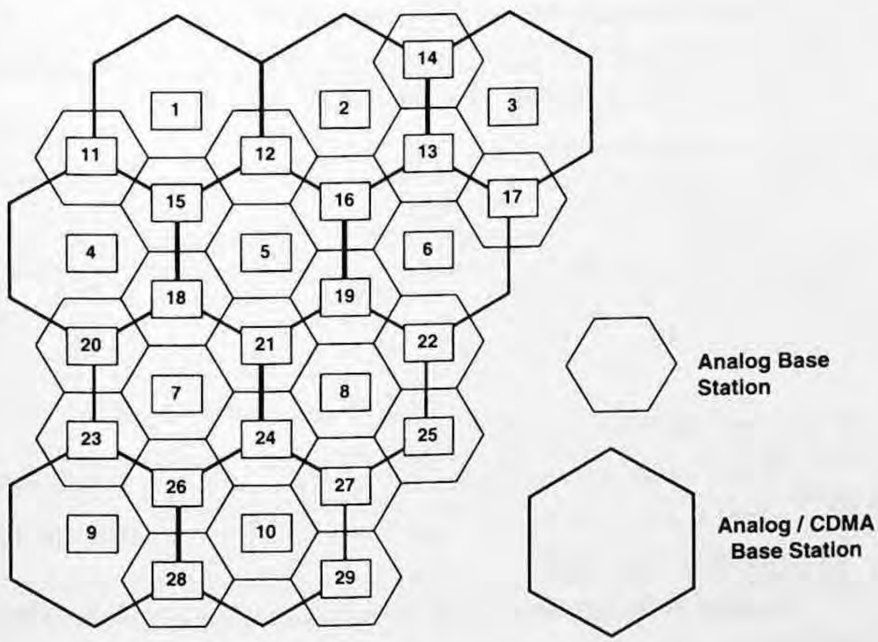

Figure 9.8 Configuration at the end of Year 1 and Year 2.

traffic load with virtually zero blocking for the CDMA subscribers. If the grade of service for the analog subscribers is an unacceptable business strategy, then additional analog capacity would have to be provided by adding channels in the limited spectrum band, possibly resulting in additional co-channel interference. The results are summarized in table 9.10. In the calculations, we assume an average of 22 CDMA calls per sector with 2 percent blocking. As can be seen, the offered CDMA traffic load (8.67 Erlangs per sector) is much lower than the 2 percent blocking capacity (14.9 Erlangs per sector). The CDMA subscribers will experience virtually no blocking, while the analog subscribers will experience 4 percent blocking.

Table 9.10 At the End of Year 1

No. of Base Stations		No. of RF Channels/Sector		Capacity, Erlangs/sector		
Analog	CDMA	Analog	CDMA	Analog	CDMA	BHCA
29	10	11	1	6.83[a]	14.9	42,000

a. 4 percent blocking.

Year 2 (see fig. 9.8 and table 9.11):

- CDMA traffic per sector: (22,000 × 0.02) / (10 × 3) = 14.67 Erlangs

One CDMA channel on 10 base stations with 2 percent blocking provides 14.9 Erlangs per sector.

- Analog traffic per sector: (24,000 × 0.02) / (29 × 3) = 5.52 Erlangs
- Number of analog channels per sector: 14 – 3 = 11

These 11 channels will carry 5.84 Erlangs of traffic at 2 percent blocking for analog subscribers. Table 9.11 summarizes the results.

Table 9.11 At the End of Year 2

No. of Base Stations		No. of RF Channels/Sector		Capacity, Erlangs/sector		
Analog	CDMA	Analog	CDMA	Analog	CDMA	BHCA
29	10	11	1	5.84	14.9	46,200

Year 3 (see fig. 9.9 and table 9.12):

- CDMA traffic per sector: (39,000 × 0.02) / (10 × 3) = 26.0 Erlangs

Two CDMA channels on 10 base stations with 2 percent blocking gives 34.68 Erlangs per sector.

- Analog traffic per sector: (16,000 × 0.02) / (23 × 3) = 4.64 Erlangs

We remove 6 analog base stations leaving 23 base stations to carry analog traffic.

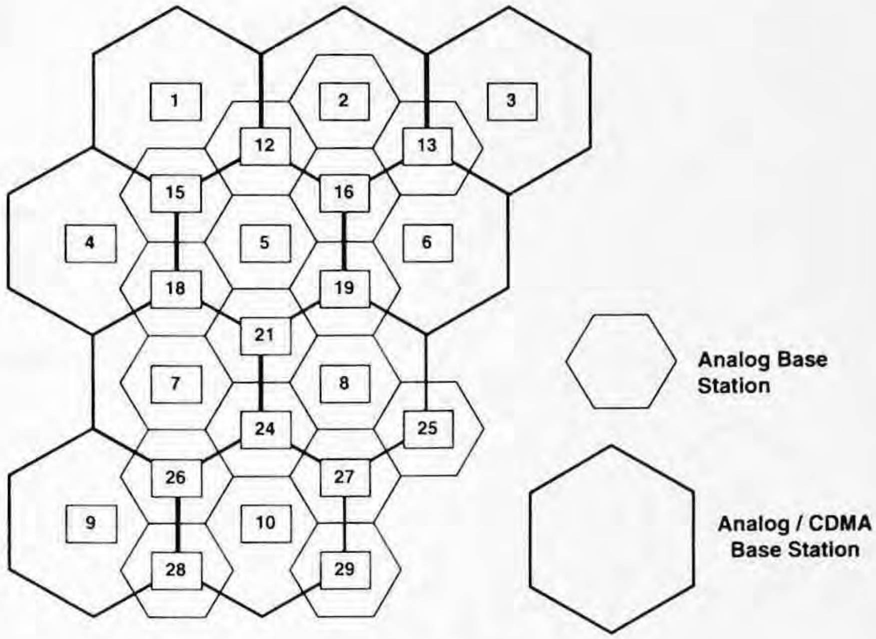

Figure 9.9 Configuration at the end of Year 3.

- Number of analog channels per sector: 14 – 5 = 9

These 9 channels will carry 4.35 Erlangs per sector of traffic at 2 percent blocking for the analog subscribers. Since the offered traffic load is slightly more than the capacity at 2 percent blocking, the analog subscribers will experience about 2.5 percent blocking, whereas the CDMA subscribers will experience almost no blocking since the offered load is less than the capacity. The results are summarized in table 9.12.

Table 9.12 At the End of Year 3

No. of Base Stations		No. of RF Channels/Sector		Capacity, Erlangs/sector		
Analog	CDMA	Analog	CDMA	Analog	CDMA	BHCA
23	10	9	2	4.35	34.68	55,000

Year 4 (see fig. 9.10 and table 9.13):

- CDMA traffic per sector: (52,000 × 0.02) / (3 × 10) = 34.67 Erlangs

Two CDMA channels on 10 base stations with 2 percent blocking gives 34.68 Erlangs per sector.

- Analog traffic per sector: (10,000 × 0.02) / (20 × 3) = 3.33 Erlangs
- Number of analog channels per sector: 14 – 5 = 9

We remove 3 analog base stations, leaving 20 base stations to carry analog traffic.

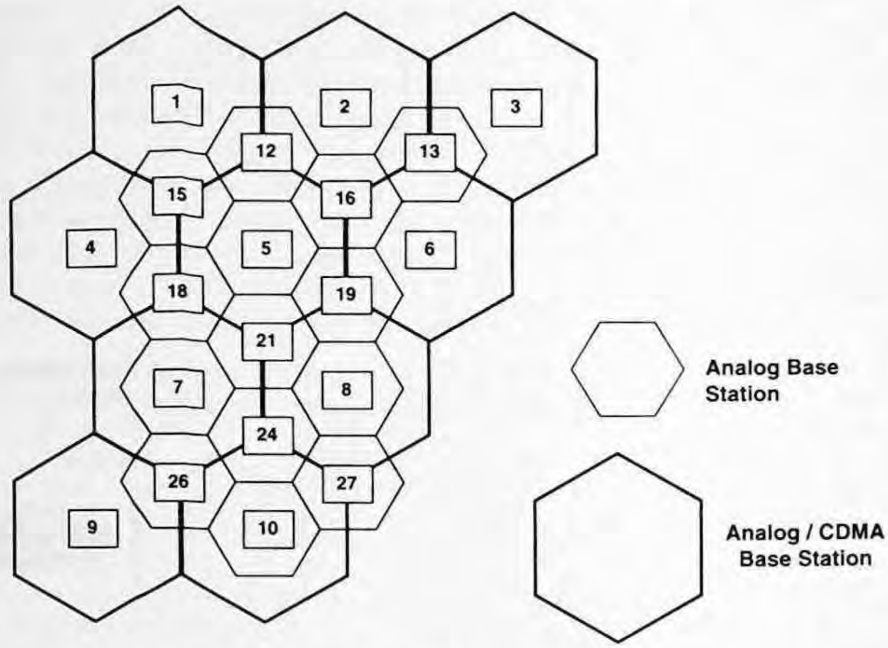

Figure 9.10 Configuration at the end of Year 4.

These 9 channels will carry 4.35 Erlangs per sector of traffic at 2 percent blocking for the analog subscribers. The results are summarized in table 9.13.

Table 9.13 At the End of Year 4

No. of Base Stations		No. of RF Channels/Sector		Capacity, Erlangs/sector		
Analog	CDMA	Analog	CDMA	Analog	CDMA	BHCA
20	10	9	2	4.35	34.68	62,000

Year 5 (see fig. 9.11 and table 9.14):

- CDMA traffic per sector: $(66{,}000 \times 0.02) / (10 \times 3) = 44.0$ Erlangs

Three CDMA channels on 10 base stations with 2 percent blocking gives 55.33 Erlangs per sector.

- Analog traffic per sector: $(4{,}000 \times 0.02) / (10 \times 3) = 2.67$ Erlangs

We remove 10 analog base stations, leaving 10 base stations to carry analog traffic.

- Number of analog channels per sector: $14 - 7 = 7$

These 7 channels will carry 2.94 Erlangs per sector at 2 percent blocking. The results are summarized in table 9.14.

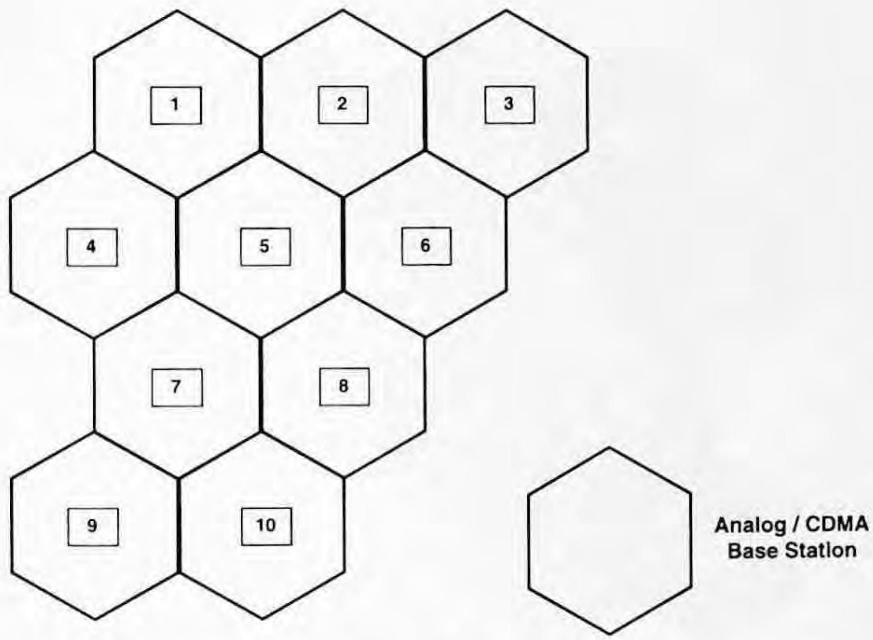

Figure 9.11 Configuration at the end of Year 5.

Table 9.14 At the End of Year 5

No. of Base Stations		No. of RF Channels/Sector		Capacity, Erlangs/sector		
Analog	CDMA	Analog	CDMA	Analog	CDMA	BHCA
10	10	7	3	2.94	55.33	70,000

Year 6 (all digital) (see fig. 9.12 and table 9.15):

- CDMA traffic per sector: $(82{,}000 \times 0.02) / (10 \times 3) = 54.67$ Erlangs

We eliminate all remaining analog base stations and use 4 CDMA channels on 10 base stations with 2 percent blocking to give 76.38 Erlangs per sector. The results are summarized in table 9.15.

Table 9.15 At the End of Year 6

No. of Base Stations		No. of RF Channels/Sector		Capacity, Erlangs/sector		
Analog	CDMA	Analog	CDMA	Analog	CDMA	BHCA
—	10	—	4	—	76.38	82,000

Year 7 (see fig. 9.12 and table 9.16):

- CDMA traffic per sector: $(100{,}000 \times 0.02) / (10 \times 3) = 66.67$ Erlangs

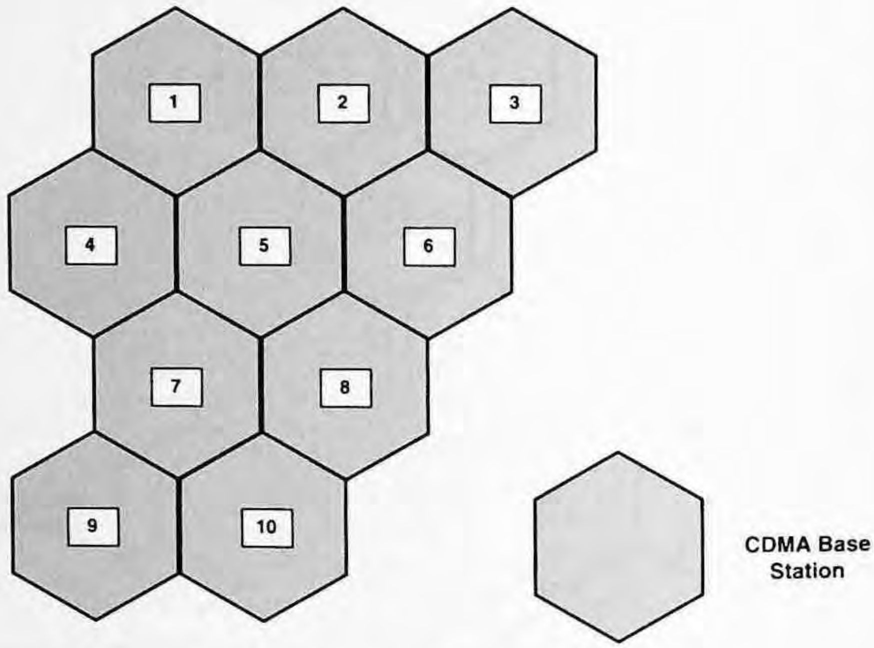

Figure 9.12 Configuration at the end of Year 6 and Year 7.

Four CDMA channels per sector provide 76.38 Erlangs of traffic at 2 percent blocking. The results are given in table 9.16.

Table 9.16 At the End of Year 7

No. of Base Stations		No. of RF Channels/Sector		Capacity, Erlangs/sector		
Analog	CDMA	Analog	CDMA	Analog	CDMA	BHCA
—	10	—	4	—	76.38	100,000

9.7 Facilities Engineering

CDMA technology offers a significant capacity improvement with respect to analog and other digital technologies, as was discussed in chapter 2. However, in order to realize these capacity improvements fully, the service provider must properly engineer the CDMA system. This section discusses the engineering of facilities and its relationship to call capacity.

Facilities encompass terrestrial facilities, radio facilities, transcoders (vocoders), and network facilities. As discussed in chapter 4, the transcoders can be physically located at the base station (see fig. 9.13) or at the mobile switching center (see fig. 9.14).

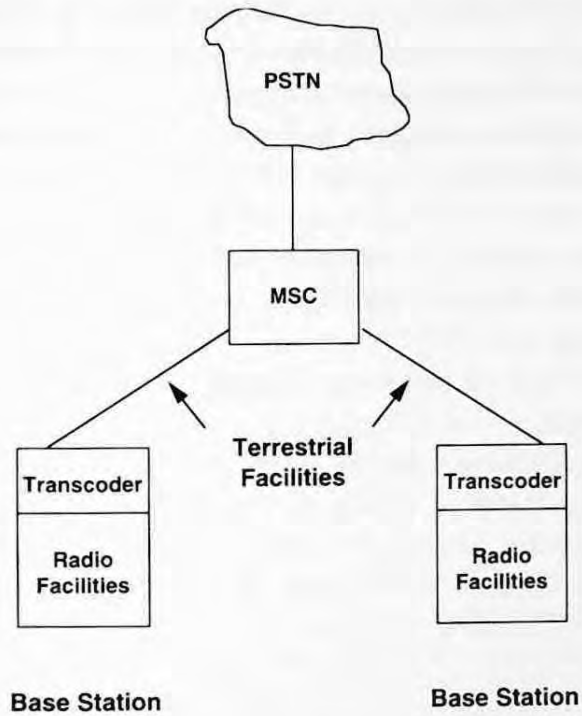

Figure 9.13 Basic CDMA facilities configuration (transcoder at base station).

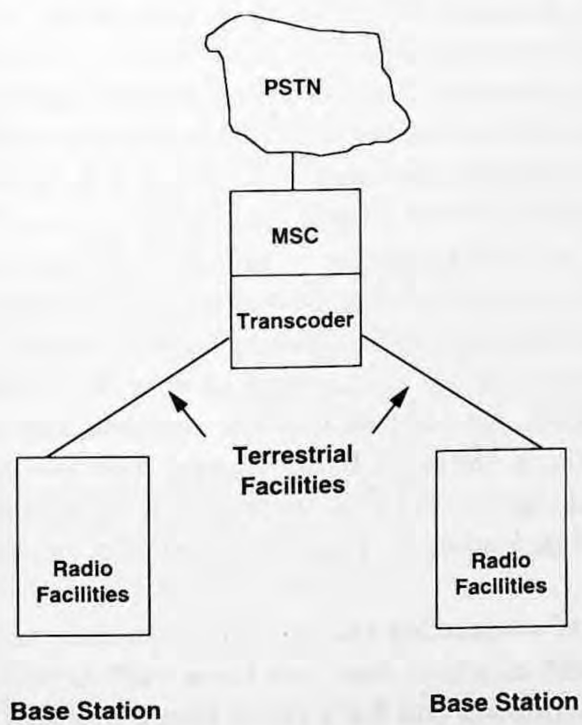

Figure 9.14 Basic CDMA facilities configuration (transcoder at MSC).

Only the radio facility is related to the capacity of the radio link. Also the service provider must configure a sufficient number of CDMA channels to support the maximum number of simultaneous calls. CDMA channels include the pilot channel, sync channel, access channels, paging channels, and traffic channels. Once this is done, the required frequency spectrum can be determined. This section provides an analytical discussion of the capacity of the reverse radio link and a qualitative discussion of the capacity of the forward radio link.

Unlike analog and TDMA technology, CDMA technology does not impose a definite limit on the radio capacity. Rather, CDMA technology exercises a *soft limit*, in which mobile subscribers experience a level of degradation that is related to the total interference and the thermal noise. The capacity is soft since the number of mobile subscribers can be increased if the service provider is willing to lower the grade of service and thus decrease customer satisfaction. With a greater number of simultaneous CDMA calls, the noise floor increases. If the noise floor increases, the probability of receiving a correct frame decreases (i.e., the frame error rate increases). Interference is generated by other CDMA mobile stations that are occupying the same radio spectrum on the same base station or on a different base station. Moreover, for 800-MHz operation, interference is generated if a mobile station that is operating in the analog mode is occupying a portion of the frequency spectrum that is used by the CDMA channel. However, in a properly engineered CDMA system, analog operation is restricted to base stations sufficiently separated from the base station serving the CDMA mobile. This significantly reduces the interference to the CDMA mobile.

In order for a CDMA system to achieve the expected capacity enhancement, it is imperative that power control be properly functioning in both the forward and reverse directions of the radio channel (refer to chapter 5). Power control is executed only on the traffic channels. However, in the following analysis, it is not assumed that power control is perfect. This fact is reflected by assuming that the instantaneous power varies about the desired E_b/I_o level with a lognormal distribution having a standard deviation σ_c. Typical values of σ_c are in the order of 1.5–2.5 dB.

One method of estimating the capacity is to determine the probability that a CDMA channel does not have sufficient bandwidth to accommodate a mobile station for a given frame interval and still satisfy the interference constraints. This event is called an *outage*. During an outage of the reverse radio channel, the frame error rate can exceed the desired maximum limit. This situation is not catastrophic but does lead to degraded service. In the following analysis, the desired interference is given by $(I_0B_w/N_0B_w) = (I_0/N_0) < (1/\eta)$. Typically, η is between 0.25 and 0.1, which corresponds to spectral density ratios of (I_0/N_0) between 6 and 10 dB.

The explicit formula for the normalized average user occupancy (λ/μ in terms of Erlangs per sector) is given by [10] and [11]:

$$\frac{\lambda}{\mu}\upsilon(1 + N_I) = K_0 \times F(B, \sigma_c) \tag{9.60}$$

where λ = average call rate for the entire CDMA system,

$1/\mu$ = average call duration,

υ = voice activity factor,

N_I = interference from base stations outside service area normalized by the interference from the base station serving the service area (in our analysis, N_I will be a constant that is dependent upon the type of CDMA handoff),

$$B = \frac{[Q^{-1}(P_{out})]^2}{K_0},$$

$$K_0 = \frac{\frac{B_w}{R}}{\frac{E_{b0}}{I_0}} \times (1 - \eta),$$

$$F(B, \sigma_c) = \frac{1}{\alpha_c}\left[1 + \frac{\alpha^3_c \cdot B}{2}\left(1 - \sqrt{1 + \frac{4}{\alpha^3_c \times B}}\right)\right], \text{ in which}$$

$$\alpha_c = e^{(\beta\sigma_c)^2/2}, \text{ and } \beta = (\ln(10))/10 = 0.2303 .$$

$Q^{-1}(z)$ is the inverse function of $Q(z)$ where

$$Q(z) = \left(\int_z^{\infty} e^{((-x^2)/2)}dx\right)/(\sqrt{2\pi}).$$

P_{out} is the probability of outage, N_I is the mean interference from neighboring base stations, and E_{b0}/I_0 is the median of the desired E_b/I_0.

Example 9.5

The following example illustrates the first entry in table 9.18. We calculate the maximum capacity of a CDMA system supporting a bandwidth of 1.25 MHz, 8-kbps vocoders, and 2-way soft handoffs. We assume that the voice activity factor equals 0.4, $E_{bo}/I_o = 2.51$ or 4 dB, $P_{out} = 0.01$, the standard deviation of the power control equals 1.78 or 2.5 dB and $(I_o/N_0) \le 10$. We determine the capacity of the reverse radio link for each antenna sector.

From the above information, we know that $\eta = 0.1$. As per TIA IS-95A and IS-96A, the reverse radio link must support 9600 bps to support the 8-kbps vocoder. From equation (9.60), we can calculate K_0:

$$K_0 = \frac{B_w/R}{E_{b0}/I_0} \times \langle 1 - \eta \rangle = \frac{1{,}250{,}000/9{,}600}{2.51} \times (1 - 0.1) = 46.69$$

$$B = \frac{[Q^{-1}(0.01)]^2}{K_0} = \frac{(2.33)^2}{46.69} = 0.116\,,\ \alpha_c = e^{(0.23 \times 1.778)^2/2} = 1.0876$$

$$F(B, \sigma_c) = \frac{1}{\alpha_c}\left[1 + \frac{\alpha^3{}_c \cdot B}{2}\left(1 - \sqrt{1 + \frac{4}{\alpha^3{}_c \times B}}\right)\right]$$

$$F(B,\sigma_c) = \frac{1}{1.0876} \times \left[1 + \frac{1.0876^3 \times 0.116}{2} \times \left(1 - \sqrt{1 + \frac{4}{1.0876^3 \times 0.116}}\right)\right] = 0.626$$

If a 2-way soft handoff is supported, $N_I = 0.77$, assuming that propagation losses increase as the fourth power of distance with a standard deviation of 8 dB. If these assumptions are not applicable, then the value of N_I may be different. In such cases, it is recommended that [10] be referenced.

$$\frac{\lambda}{\mu} = \frac{K_0 \times F(B,\sigma_c)}{\upsilon \times (1 + N_I)} = \frac{46.69 \times 0.626}{0.4 \times \langle 1 + 0.77 \rangle} = 41.3 \text{ Erlangs/sector}$$

Using equation (9.60), the following configurations are analyzed. It is assumed that attenuation due to propagation losses decrease as the fourth power of distance and that the lognormal component has a standard deviation of 8 dB. Refer to equation (9.1). The voice activity factor υ is assumed to equal 0.4, and the standard deviation for power control σ_c is assumed to be 2.5 dB. In tables 9.17 to 9.25 we use $N_I = 2.38$ for hard handoffs, $N_I = 0.77$ for soft handoffs with a maximum of 2 base stations, and $N_I = 0.57$ for soft handoffs with a maximum of 3 base stations.

1. Hard handoff only for a CDMA system with a frequency bandwidth of 1.25 MHz and equipped with 8-kbps transcoders (see table 9.17), $B_w = 1.25$ MHz, $R = 9.6$ kbps, and $B_w/R = G_p = 130$;

where G_p is the processing gain, B_w is the bandwidth of the channel, and R is the information rate.

Table 9.17 Configuration 1—Hard Handoffs Only (N_I = 2.38), Bandwidth = 1.25 MHz, 8-kbps Transcoders (B_w/R = 130)

η	E_{b0}/I_0	P_{out}	λ/μ, Erlangs
0.10	2.51 (4 dB)	0.01	21.63
0.25	2.51	0.01	17.38
0.10	3.98 (6 dB)	0.01	12.38
0.25	3.98	0.01	9.86
0.10	2.51	0.05	24.18
0.25	2.51	0.05	19.63
0.10	3.98	0.05	14.23
0.25	3.98	0.05	11.48

2. Soft handoffs with a maximum of two base stations with a 1.25-MHz bandwidth and equipped with 8-kbps transcoders (see table 9.18).

Table 9.18 Configuration 2—Soft Handoffs (N_I = 0.77), Bandwidth = 1.25 MHz, 8-kbps Transcoders (B_w/R = 130)

η	E_{b0}/I_0	P_{out}	λ/μ, Erlangs
0.10	2.51 (4 dB)	0.01	41.30
0.25	2.51	0.01	33.19
0.10	3.98 (6 dB)	0.01	23.63
0.25	3.98	0.01	18.83
0.10	2.51	0.05	46.17
0.25	2.51	0.05	37.49
0.10	3.98	0.05	27.17
0.25	3.98	0.05	21.92

3. Soft handoffs with a maximum of three base stations with a 1.25-MHz bandwidth and equipped with 8-kbps transcoders (see table 9.19).

Table 9.19 Configuration 3—Soft Handoffs (N_i = 0.57), Bandwidth = 1.25 MHz, 8-kbps Transcoders (B_w/R = 130)

η	E_{b0}/I_0	P_{out}	λ/μ, Erlangs
0.10	2.51 (4 dB)	0.01	46.56
0.25	2.51	0.01	37.42
0.10	3.98 (6 dB)	0.01	26.64
0.25	3.98	0.01	21.23
0.10	2.51	0.05	52.05
0.25	2.51	0.05	42.27
0.10	3.98	0.05	30.63
0.25	3.98	0.05	24.71

4. Hard handoffs only with a 1.25-MHz bandwidth and equipped with 13-kbps transcoders (see table 9.20), B_w = 1.25 MHz, R = 14.4 kbps, and B_w/R = 87.

Table 9.20 Configuration 4—Hard Handoffs Only (N_i = 2.38), Bandwidth = 1.25 MHz, 13-kbps Transcoders (B_w/R = 87)

η	E_{b0}/I_0	P_{out}	λ/μ, Erlangs
0.10	2.51 (4 dB)	0.01	13.25
0.25	2.51	0.01	10.46
0.10	3.98 (6 dB)	0.01	7.43
0.25	3.98	0.01	5.86
0.10	2.51	0.05	15.17
0.255	2.51	0.05	12.25
0.10	3.98	0.05	8.79
0.25	3.98	0.05	7.04

5. Soft handoffs with a maximum of two base stations with a 1.25-MHz bandwidth and equipped with 13-kbps transcoders (see table 9.21).
6. Soft handoffs with a maximum of three base stations with a 1.25-MHz bandwidth and equipped with 13-kbps transcoders (see table 9.22).

Table 9.21 Configuration 5—Soft Handoffs (N_i = 0.77), Bandwidth = 1.25 MHz, 13-kbps Transcoders (B_w/R = 87)

η	E_{b0}/I_0	P_{out}	λ/μ, Erlangs
0.10	2.51 (4 dB)	0.01	25.29
0.25	2.51	0.01	20.17
0.10	3.98 (6 dB)	0.01	14.18
0.25	3.98	0.01	11.19
0.10	2.51	0.05	28.97
0.255	2.51	0.05	23.39
0.10	3.98	0.05	16.79
0.25	3.98	0.05	13.45

Table 9.22 Configuration 6—Soft Handoffs (N_i = 0,57), Bandwidth = 1.25 MHz, 13-kbps Transcoders (B_w/R = 87)

η	E_{b0}/I_0	P_{out}	λ/μ, Erlangs
0.10	2.51 (4 dB)	0.01	28.52
0.25	2.51	0.01	22.74
0.10	3.98 (6 dB)	0.01	15.99
0.25	3.98	0.01	12.62
0.10	2.51	0.05	32.66
0.255	2.51	0.05	26.37
0.10	3.98	0.05	18.93
0.25	3.98	0.05	15.17

7. Hard handoffs only with a 10-MHz bandwidth and equipped with 8-kbps transcoders (see table 9.23) B_w = 10 MHz, R = 9.6 kbps, and B_w/R = 1042.
8. Soft handoff with a maximum of two base stations with a bandwidth of 10 MHz and equipped with 8-kbps transcoders (see table 9.24).
9. Soft handoffs with a maximum of three base stations with a 10-MHz bandwidth and equipped with 8-kbps transcoders (see table 9.25).

Table 9.23 Configuration 7—Hard Handoffs Only (N_I = 2.38), Bandwidth = 10 MHz, 8-kbps Transcoders (B_w/R = 1042)

η	E_{b0}/I_0	P_{out}	λ/μ, Erlangs
0.10	2.51 (4 dB)	0.01	221.51
0.25	2.51	0.01	182.21
0.10	3.98 (6 dB)	0.01	134.93
0.25	3.98	0.01	110.62
0.10	2.51	0.05	230.52
0.255	2.51	0.05	190.34
0.10	3.98	0.05	141.87
0.25	3.98	0.05	116.86

Table 9.24 Configuration 8—Soft Handoffs (N_I = 0.77), Bandwidth = 10 MHz, 8-kbps Transcoders (B_w/R = 1042)

η	E_{b0}/I_0	P_{out}	λ/μ, Erlangs
0.10	2.51 (4 dB)	0.01	423.00
0.25	2.51	0.01	347.93
0.10	3.98 (6 dB)	0.01	257.66
0.25	3.98	0.01	211.24
0.10	2.51	0.05	440.21
0.255	2.51	0.05	363.48
0.10	3.98	0.05	270.91
0.25	3.98	0.05	223.16

Table 9.25 Configuration 9—Soft Handoffs (N_I = 0.57), Bandwidth = 10 MHz, 8-kbps Transcoders (B_w/R = 1042)

η	E_{b0}/I_0	P_{out}	λ/μ, Erlangs
0.10	2.51 (4 dB)	0.01	476.88
0.25	2.51	0.01	392.28
0.10	3.98 (6 dB)	0.01	290.48
0.25	3.98	0.01	238.15
0.10	2.51	0.05	496.29
0.255	2.51	0.05	409.79
0.10	3.98	0.05	305.42
0.25	3.98	0.05	251.59

Configurations 1, 2, and 3 correspond to wireless systems supporting standards TIA IS-95A and TIA IS-96A; configurations 4, 5, and 6 correspond to wireless systems supporting standards TIA IS-95A and Qualcomm's proprietary 13-kbps vocoder; configurations 7, 8, and 9 correspond to TIA SP-2977 and TIA IS-96A.

Comparing tables 9.17 to 9.25, we can make several observations:

- Hard handoffs reduce the capacity on the reverse radio link by approximately 50 percent with respect to two-way soft handoffs (table 9.17 vis-à-vis table 9.18).
- If a system is equipped with 13-kbps transcoders rather than 8-kbps transcoders, the capacity is reduced approximately 40 percent (table 9.19 vis-à-vis table 9.22). This observation is consistent with our expectations since the rate is increased from 9.6 to 14.4 kbps (50 percent increase).
- If the bandwidth increases from 1.25 to 10 MHz (8 times increase), the capacity increases approximately 10 times (table 9.18 vis-à-vis table 9.24).

We investigate the effect on the reverse radio link capacity if the power control is perfect (i.e., $\sigma_c = 0$). In this case, $F(B, \sigma_c)$ in equation (9.60) simplifies to:

$$F(B, \sigma_c) = 1 + \frac{B}{2}\left(1 - \sqrt{1 + \frac{4}{B}}\right) \tag{9.61}$$

As an example of applying equation (9.61), we can compare the results in table 9.18 in which $\eta = 0.10$, $(E_{b0}/I_0) = 6\text{dB}$, and $P_{out} = 0.01$. In this case, $(\lambda/\mu) = 23.63$ Erlangs. If the power control is perfect (i.e., $\sigma_c = 0$), (λ/μ) increases to 27.14 Erlangs or a 15 percent increase. However, equation (9.61) is based upon equation (9.1), which may not adequately model all radio environments. In such cases, the engineer needs to use either a more complicated mathematical model or to execute a computer simulation.

When determining the capacity of the reverse radio link, one must also include the capacity needed to support the access channels. Both call setup and registration messages are transmitted on the access channel. The decrease of the capacity on the reverse radio link is small due to supporting the access channels, typically about 1 percent reduction of the supportable Erlang traffic. Thus, this reduction is ignored in the determination of the radio link capacity.

As discussed in chapter 7, a call may be simultaneously supported by multiple base stations on the traffic channel. This configuration is called a *soft handoff*. A call may be simultaneously supported by multiple sectors on the same base station. This configuration is called a *softer handoff*. Moveover, a call may be in both a soft handoff and a softer handoff at a particular instance of time. This configuration is called a *soft-softer handoff* (see fig. 9.15). In this example, the mobile station is communicating with sectors D and F (softer handoff with the first base station) and with sector B (in the second base station). Thus, the mobile station is also in soft handoff with both base stations.

When designing a CDMA system for a given call load, the engineer must determine the number of CDMA channel modem (CM) circuits that must be supported at each base station. A CM operates at baseband rather than at RF frequencies. It demodulates the CDMA signal for a given mobile station and combines signals from multiple sectors of a given base station during softer handoffs. The number of CMs is affected by the number of simultaneous soft handoffs but not upon softer handoffs. In the analysis, it is assumed that a CM is not dedicated to a particular sector or CDMA carrier, although this assumption is dependent upon the actual manufacturer's implementation.

The exact distribution of two-way soft handoffs, three-way soft handoffs, softer handoffs, and soft-softer handoffs is very dependent upon the radio environment and radio configuration. The service provider needs to tune the system in order to optimize the radio link capacity. This tuning process includes adjusting the transmitted power of the

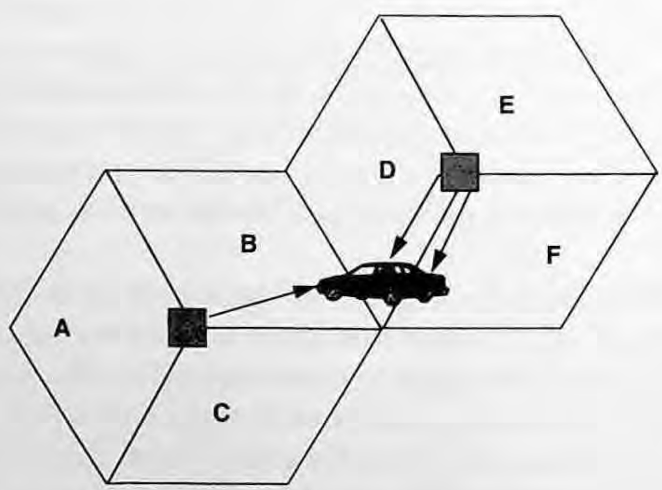

Figure 9.15 Soft-softer handoff configuration.

pilot channels and the threshold levels that trigger handoffs. TIA IS-95A defines several thresholds: T_ADD, T_DROP T_TDROP, and T_COMP. The base station sends the values of these thresholds in the *systems parameter message* (which is transmitted on the paging channel) and the *handoff direction message* or the *extended handoff direction message* (which is transmitted on the traffic channel). During a call, the mobile station measures the strength of the pilot channel of the serving base stations (sectors) and potential candidates. The mobile station sends a *pilot strength measurement message* to the base station in order to initiate a possible handoff for one of the following reasons:

- Pilot's signal strength of a serving cell drops below T_DROP for a duration equal to T_TDROP.[7] T_TDROP is determined by the service provider and ranges from 0 to 319 seconds.
- Pilot's signal strength of a candidate exceeds T_ADD.[8]
- Pilot's signal strength of a candidate exceeds that of a serving base station by T_COMP.[9]

As an example, table 9.26 shows the handoff distribution exhibited during CDMA trials in San Diego, California.

Table 9.26 Handoff Distribution—CDMA Formal Field Test, November 18–23, 1991

Handoff Type	Mean, percent	Standard Deviation, percent
Softer	62.80	19.14
Two-way soft	6.52	19.05
Soft-softer	4.42	7.90
Three-way soft	0.40	0.82
Three-way softer	0.20	0.42
No handoff	25.66	10.80

As a general rule, a CDMA system is tuned so that calls are in some form of soft handoff (i.e., two-way soft, soft-softer, or three-way soft) for

7. As defined in Section 6.6.6.2.5.2 of TIA IS-95A, this pilot is contained in the Active Set.

8. TIA IS-95A refers to the pilot as being in the Neighbor Set or in the Remaining Neighbor Set.

9. TIA IS-95A refers to the pilot as being in the Candidate Set.

approximately 30 percent of the time. Percentages greater than this often are not justified by the improvement of the call quality. It is interesting to note that the aggregated results of the CDMA Formal Field Test (table 9.26) indicate that the total soft handoff (the sum of two-way soft, three-way soft, and soft-softer) is 12 percent of the time. This observation may be rationalized by the fact that calls were in total softer handoff (the sum of softer, soft-softer, or three-way softer) 67 percent of the time. Also, it should be noted that the standard deviation of the handoff distributions is large, particularly for two-way soft and soft-softer handoffs. There are two reasons for this:

- The handoff distribution varies with the base stations that are serving the call. Base stations are tuned differently in order to optimize the call capacity.
- The handoff distribution varies with the path of the mobile station. Even though the mobile station may be served by the same base station, the terrain within the base station's domain varies sufficiently to significantly affect the handoff distributions.

When a call is in a two-way soft, soft-softer, or three-way softer handoff, two CMs are required to support the call. When a call is in a three-way soft handoff, three CMs are needed. Only one CM is needed for the call during a softer handoff or when no handoff configuration occurs.

Example 9.6

Assume an equal call distribution across all base stations and sectors with the same handoff distribution. The handoff distribution follows: 40 percent softer handoff, 20 percent two-way soft handoff, 10 percent soft-softer handoff, 29 percent no handoff, and 1 percent three-way soft handoff. There are two CDMA carriers, each having a bandwidth of 1.25 MHz. The system is equipped with transcoders that conform to TIA IS-96A (i.e., 8-kbps voice coding and 9.6 kbps on the physical layer). Assume $\eta = 0.25$, $E_{b0}/I_0 = 6$ dB, and $P_{out} = 0.01$. The system is configured with 10 base stations, each having three sectors. The average call duration is 90 seconds, and each mobile subscriber generates 0.03 Erlang during the busy hour.

Determine the number of calls that can be supported per hour by the system. Determine the number of mobile subscribers that can be provided service if 2 percent blocking is acceptable. Also, determine the number of CMs that must be equipped to support the calculated number of subscribers.

From table 9.19, the capacity of the reverse radio link per sector is:

$$21.23 \ \frac{\text{Erlangs}}{\text{carriers}} \times 2 \text{ carriers} = 42.46 \text{ Erlangs}$$

Thus, each sector can simultaneously support 42 CDMA channels. This does not equal the number of simultaneous calls because some of the channels are assigned as the second and third channels for calls in softer and soft handoff. A CDMA channel is required for each sector configured in the call. Refer to figure 9.16.

In order to determine the number of simultaneous calls that can be supported by the CDMA system, we must associate each call with one sector, even though the given call is being served by multiple sectors. In this example, we assume that the call is associated with the oldest serving sector, although other assignments may be assumed. If complete homogeneity is assumed, we can determine the total number of simultaneous calls supported by the entire system by multiplying the number of sectors ($10 \times 3 = 30$) with the number of simultaneous calls supported by each sector. To illustrate this point, let us determine the number of simultaneous calls supported by sector B as shown in figure 9.16. Softer handoffs are served by sectors B-C and B-A; two-way soft handoffs, by B-G and B-F; three-way soft handoffs, by B-G-F; soft-softer handoffs, by B-C-G, B-C-J, B-C-F, B-C-N, B-A-U, B-A-F, B-A-Q, and B-A-G; no handoffs, by B. Let the number of simultaneous calls supported by the sector be x. Then x can be determined by

$$x + 0.40x + 0.2x + 0.1x + 0.01x = 42$$

where 40 percent of the channels are supporting other sectors in softer handoff, 20 percent of the channels are supporting other base stations in two-way soft handoff, 10 percent of the channels are supporting other base stations in soft-softer handoff, and 1 percent of the channels are supporting base stations in three-way soft handoff.

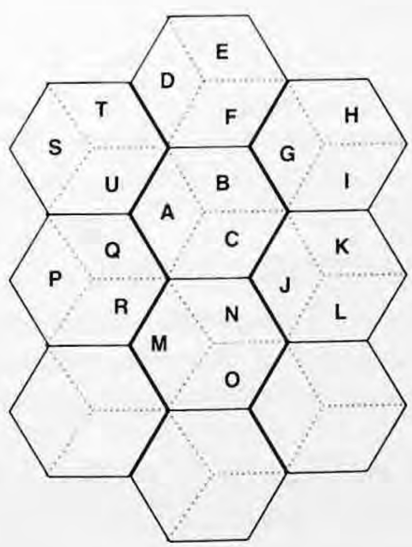

Figure 9.16 Sector configuration for Example 9.6

Thus, x equals 24 simultaneous calls per sector, and the entire CDMA system can serve:

$$10 \text{ cells} \times 3 \frac{\text{sectors}}{\text{base station}} \times 24 \frac{\text{calls}}{\text{sector}} = 720 \text{ calls}$$

In order to determine the number of subscribers that can be supported by this system by each sector during the busy hour, we use the Erlang-B formula. Each sector can support 24 simultaneous calls, which is equivalent to trunks or radios. From Erlang-B tables, we find that 16.63 Erlangs per sector can be supported during the busy hour. Thus, the number of mobile subscribers that the system supports is determined by

$$0.03 \frac{\text{Erlangs}}{\text{subscriber}} \times N(\text{subscribers}) = 16.63 \frac{\text{Erlangs}}{\text{sector}} \times 3 \frac{\text{sectors}}{\text{base station}} \times 10 \text{ base station}$$

$$N = \frac{16.63 \times 3 \times 10}{0.03} \text{ subscribers}$$

$$N = 16630 \text{ subscribers}$$

Next, we calculate the number of CMs that need to be equipped at each sector. We have already determined that each sector can support 42 CDMA channels. If a call is in a softer or no handoff, one CM is needed; for two-way soft and soft-softer handoff, two CMs are needed; for a three-way soft handoff, three CMs are needed. The number of CMs that need to be equipped at each sector for supporting the traffic channels is

$$y + 0.2y + 0.1y + 0.01y = 42 \text{ CM}$$

where 20 percent of the CMs are supporting other base stations in soft handoff, 10 percent of the CMs are supporting other base stations in soft-softer handoff, and 1 percent of the CMs are supporting other base stations in three-way soft handoff.

$$y = 32 \text{ CM}$$

In addition, CMs must be equipped for the access channel. Even though the access channel has a negligible effect upon the reverse radio channel ($16.31 \times 0.01 = 0.02$ Erlang), a CM must be equipped to support this channel.

9.8 Capacity of Forward Radio Channel

We have assumed that the reverse radio channel is the limiting factor when determining the capacity of the radio channel. Qualitatively, we can justify this assumption by the following observations:

- As per TIA IS-95A, the forward radio channel is orthogonal, while the reverse radio channel is approximately orthogonal.
- Interference is from a few sources (base stations) to many receivers (mobile stations) on the forward radio channel, while the interfer-

ence is from many sources (mobile stations) to a few receivers (base stations) on the reverse radio channel.

- The pilot channel supports synchronization on the forward radio channel.

9.9 Design Considerations at the Boundary of a CDMA System

Standards do not currently support soft handoffs between CDMA systems that are operated by different service providers. Thus, if a mobile station moves between these CDMA systems, a hard handoff will occur. This results in the reduction of capacity. Comparing table 9.17 with table 9.18, we see that the capacity is degraded by approximately 50 percent. The service provider has several options in order to mitigate this problem. First, the boundary base stations (sectors) can be located at low traffic areas. If this is not practical, additional spectra can be allocated at the boundary sectors. Neither option is very appealing to the service provider. Consequently, TIA TR-45 is currently developing an intervendor soft handoff.

9.10 Summary

This chapter discusses principles for engineering a CDMA system including propagation models, link budgets, and facilities engineering. However, extensive RF measurements and RF modeling are needed to plan a real commercial system. The intent of this chapter is to provide some tools for a better understanding of achieving this goal.

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CHAPTER 10

Wireless Data

10.1 Introduction

In this chapter, we discuss wireless data systems including the wide area wireless data systems and the high-speed wireless local area networks. We present activities for wireless data standards and outline the error control methods used by the standards. We also cover packet radio protocols and their channel efficiency formulas. The contention function of packet radio models the mechanism where mobile stations access the network on the access channel. Packet services are one of the four data services that are supported in CDMA. The other three are asynchronous data, facsimile, and short message services (similar to paging), which are also covered in this chapter.

We discuss the standards for data services that are supported by CDMA cellular/personal communications systems and present highlights of the TIA IS-99, TIA IS-637, and TIA IS-657 standards. We describe the architecture for each of the four data services and the protocol stacks that are supported by the services.

We include both sets of standards (CDMA and non-CDMA) for two reasons:

- The WLANs all use some form of spread spectrum communications, either frequency hopping or direct sequence spreading.
- The two methods (WLANs and CDMA) are part of a larger wireless network that many companies are constructing.

With the phenomenal growth of laptop personal computers and the Internet, wireless data are no longer limited to just e-mail or faxes. It encompasses the ability to send and receive data anytime, from any place in the world, and to provide to a user, at a remote location, full access to all of their desktop services that would normally be available at their office PC. Data services are delivering the same promise that voice services have recently delivered: anytime, anywhere communications.

10.2 Wireless Data Systems

We can classify wireless data systems into two basic categories: wide-area wireless data systems and high-speed WLANs. WLANs and wide area wireless data systems serve different categories of user applications and, therefore, have different system design objectives. Wireless data services are used for transaction processing and for interactive, broadcast, and multicast services. Transaction processing is used for credit card verification, paging, taxi calls, vehicle theft reporting, and notice of voice or electronic mail. Interactive services include database access and remote LAN access. Broadcast services are general information services, weather and traffic advisory services, and advertising. Multicast services are similar to subscribed information services, law enforcement communications, and private bulletin boards.

In the following sections, we briefly describe wide area wireless data systems and WLANs that have been deployed in the United States.

10.2.1 Wide Area Wireless Data Systems

Wide area wireless data systems are designed to provide high mobility, wide area coverage, and low data rate digital data communications to both vehicles and pedestrians. The technical challenge is to design a system that efficiently uses the available bandwidth to serve large numbers of users distributed over wide geographical areas. Table 10.1 gives the details of the wide area wireless packet data systems that have been deployed in the United States. Specialized mobile radio services (SMRS) allocations are centered around 450 MHz and 900 MHz in the United States.

The ARDIS data network was developed by Motorola as a joint venture between Motorola and IBM to support IBM field service repair people. It is now a public service offering and is solely owned by Motorola. RAM Mobile Data is another public offering that uses the Ericsson Mobitex technology. Both the ARDIS and RAM networks are evolving to data

Table 10.1 Wide-Area Wireless Packet Data Systems

	RAM Mobile (Mobitex)	ARDIS (KDT)	Metricom (MDN)	CDPD
Data Rate	19.2 kbps	19.2 kbps	76 kbps	19.2 kbps
Channel Spacing	12.5 kHz	25 kHz	160 kHz	30 kHz
Access	Slotted ALOHA CSMA	CSMA/CD	FHSS (ISM)	Unused AMPS channels
Frequency, MHz	$f_c \sim 900$	$f_c \sim 800$	$f_c \sim 915$	$f_c \sim 800$
Transmit Power, W	0.16–10 under power control	40	1	1.6
Modulation	GMSK	GMSK	GMSK	GMSK, BT = 0.5

rates of 19.2 kbps. They have been designed to use standard, two-way voice, land mobile-radio channels, with 12.5- or 25-kHz channel spacing.

The CDPD technology shares the 30-kHz spaced, 800-MHz voice channels used by the AMPS systems. The data rate is 19.2 kbps. The CDPD base station equipments share cell sites with the voice cellular radio system. The aim is to reduce the cost of providing packet data service by sharing the resources with the voice cellular systems. This strategy is similar to one that has been used by nationwide fixed wireline packet data networks to provide an economically viable data service by using a small portion of the capacity of the networks designed mainly for voice traffic.

Another approach in wide area wireless packet data networks is based on the microcell concept to provide coverage in smaller areas. The microcell data networks are designed for stationary or low-speed users. The basic aim is to reduce the cost of providing wireless data service by using small and inexpensive base stations that can be installed on the utility poles, the sides of buildings, and inside buildings. The strategy is similar to the one being proposed for personal communications networks. Base station-to-base station wireless links are used to reduce the cost of interconnecting a data network. A large microcell network of small inexpensive base stations has been installed in the lower San Francisco Bay Area by Metricom. The slow frequency hopping-spread spectrum in 902- to 928-MHz U.S. industrial scientific medical (ISM) band is used where transmitter power is 1 W maximum. Power control is used to minimize interference and maximize battery lifetime.

10.2.2 High-Speed Wireless Local Area Networks

A WLAN typically supports a limited number of users in a well-defined indoor area. System aspects such as bandwidth efficiency and product standardization are not crucial. The maximum achievable data rate is an important consideration in the selection of a WLAN. The transmission channel characteristics and signal-processing techniques are important.

WLANs are used to extend wired LANs for convenience and mobility. Three different approaches for connectivity of WLANs have been used. The first approach includes the access to the wide area networks (WANs) and metropolitan area networks (MANs). In the wide area, the network transmission systems use the cellular arrangement and the wired long-distance network. The data are packetized to meet the immediate demands of the users' community. A proper form and format of the data are required to prevent excessive overhead and consequent latency in transport. The second approach deals with localized communications services for the added convenience of connections between the building floors and desktop in a dynamic environment. Flexibility to provide quick connections for moves, adds, and changes gives the organization significant improvement over the basic wired LAN. The third approach is the flexible mobile LAN arrangement. This form of connectivity is becoming important in all walks of life and business communities of interest. As the workforce becomes more mobile, the need to provide untethered connectivity is increasing exponentially.

Two different technical approaches exist with the WLANs. These are based on radio and optical technologies. In the radio-based technology, there are two solutions: the licensed microwave radio frequency range (18–23 GHz) or the unlicensed radio frequency range (902–928 MHz, 2.4–2.4835 GHz, and 5.75–5.825 GHz). In the unlicensed radio frequency, there are two options. The first option uses an FHSS technology, whereas the second option uses a DSSS technology. The 902- to 928-MHz frequency band is an unlicensed ISM band that allows manufacturers to supply products with very limited constraints. Newer products are also emerging that use the 2.4-GHz band. The following are the major limitations of the unlicensed frequency band WLANs:

- The system is restricted to 100-mW output.
- The system must not interfere with other radio frequency equipment in the same area.

- The system must go through an FCC-type acceptance process (in the international sector, this is called homologation or type acceptance, and the frequencies may be different, using either 902- to 928-MHz, 2.4- to 2.4835-GHz, or 5.75- to 5.825-GHz frequencies in various ISM bands).

Spread-Spectrum Radio-Based WLANs WLANs use spread spectrum techniques to allow flexibility and minimize interference, while not being license-bound. WLANs using FHSS and DSSS approaches with different speeds have been produced. The motivation to use spread spectrum for packet radio systems comes from improved multipath resistance, the ability to coexist with other systems, and the antijamming nature of the code. In an office environment, spread spectrum is a promising choice because it reduces the effects of multipath caused by reflections from the walls and increases the mobility of the terminals within the office environment. The low spectral power density per user of spread spectrum permits an overlay with certain existing systems and reduces the concerns about the health-related issues in high-power transmission. Spread spectrum offers the potential for greater range and higher data rates compared with the optical technology. It improves interception resistance and provides data privacy.

Table 10.2 provides a partial list of WLANs available in the United States.

The following sections describe briefly two WLANs: AT&T WaveLAN based on the FHSS technology and Telesystems advanced radio LAN (ARLAN) based on the DSSS technology [1].

AT&T WaveLAN. This system supports speeds up to 20 Mbps and works with various network operating systems. WaveLAN uses a DS quadrature phase-shift keying multiplexing scheme to transmit across the entire ISM band at high signal rates. Through multiplication of the original narrow band signal with the PN-sequence, the code is spread across several frequencies. WaveLAN offers better security, because the conventional radio receiver cannot decode the signal without knowing the actual spreading pattern. WaveLAN can operate up to 800 feet with a power output of 250 mW. It works in any laptop, notebook, or palmtop PC that is equipped for Personal Computer Memory Card International Association (PCMCIA) card. WaveLAN allows users to operate in a cellular network for LANs. Each WaveLAN is assigned its own identification code and can receive data only if its code corresponds to that of the base

Table 10.2 Partial List of WLAN Products

Product	Frequency	Link Rate	User Rate	Protocol	Access	No. of Channel or Spread Factor	Mod/Coding	Power, mW	Network Topology
Altair Plus Motorola	18–19 GHz	15 Mbps	5.7 Mbps	Ethernet	—	—	4-level FSK	25 peak	8 devices per radio
WaveLAN AT&T	902–928 MHz	2 Mbps	1.6 Mbps	Ethernet-like	DSSS	—	DQPSK	250	Peer-to-peer
AirLAN Solectek	902–928 MHz	—	2 Mbps	Ethernet	DSSS	—	DQPSK	250	PCM-CIA with antenna
Freeport Windata	902–928 MHz	16 Mbps	5.7 Mbps	Ethernet	DSSS	32 chips per bit	16 PSK/ trellis	650	Hub
Intersect Persoft, Inc.	902–928 MHz	—	2 Mbps	Ethernet; Token Ring	DSSS	—	DQPSK	250	Hub
LAWN O'Neill Comm.	902—928 MHz	—	38.4 kbps	AX.25	SS	20 users per channel; max. 4 channels	—	20	Peer-to-peer
WiLan WiLan, Inc.	902–928 MHz	20 Mbps	1.5 Mbps per channel	Ethernet; Token Ring	CDMA/ TDMA	3 channels, 10–15 links each	Unconven-tional	30	Peer-to-peer
Radio Port ALPS Electric	902–928 MHz	—	242 kbps	Ethernet	SS	—	—	100	Peer-to-peer

Table 10.2 Partial List of WLAN Products (Continued)

Product	Frequency	Link Rate	User Rate	Protocol	Access	No. of Channel or Spread Factor	Mod/Coding	Power, mW	Network Topology
ARLAN 600 Telesystem	902–928 MHz, 2.4 GHz	—	1.35 Mbps	Ethernet	FHSS	—	—	1000 Max.	PCs with antenna
Radio Link Cal. Microwave	902–928 MHz, 2.4 GHz	250 kbps	64 kbps	—	FHSS	250 ms/hop 500 kHz space	—	—	Hub
RangeLAN Proxim, Inc.	902–928 MHz	—	242 kbps	Ethernet; Token Ring	DSSS	3 channel	—	100	—
RangeLAN 2 Proxim, Inc.	2.4 GHz	1.6 Mbps	50 kbps	Ethernet; Token Ring	FHSS	10 channels @ 5 kbps; 15 sub-channel each	—	100	Peer-to-peer bridge
Netwave Xircom	2.4 GHz	1 Mbps per adopter	—	Ethernet; Token Ring	FHSS	82 1-MHz chan-nel or hops	—	—	Hub
Freelink Cabletron System	2.4 and 5.8 GHz	—	5.7 Mbps	Ethernet	DSSS	32 chips per bit	16 PSK trellis	100	100

station it occupies. Users can move anywhere within their assigned base station and still be able to communicate within the base station. If users need to move between base stations, they must first stop the application running and then reconfigure their address ID to match with the cell they are moving into. With roaming, this is automatic. WaveLAN is capable of interfacing directly to the backbone cable systems at standard LAN cable speeds.

Telesystems Advanced Radio LAN. ARLAN uses DSSS technology. Using a conventional cable system, ARLAN devices, called *access points*, are attached to the cable to allow for a full range of interconnections. A microcell can be configured from the backbone network by setting an access point to act like a wireless repeater. Telesystems microcellular architecture (TMA) allows the network to cover various applications and various-sized facilities. With multiple base station antennas, the network can be extended to create microcells, each with its own operating area and devices. TMA is supported by firmware in each of the ARLAN devices and supports multiple overlapping base stations creating a seamless network within the building. Handoff from base station to base station is a part of the network concept that allows for LAN connectivity of users who need to move freely throughout departments or floors within the building. Using the SS technology, the system can select various center frequencies and allows for the coexistence of multiple devices operating within the same area serving different needs. ARLAN 600 was designed for high-noise industrial applications and uses a spreading ratio of up to 100. It offers a full range of interfaces for asynchronous and synchronous data transfer from terminals and hosts. The system operates in the 915-MHz and 2.4-GHz frequency ranges and uses packet burst duplex transmission capabilities. Access to the ARLAN network is packet-switched carrier-sensed multiple access with collision avoidance (CSMA/CA) (see section 10.4 for details on CSMA). Power output for these devices is up to 1 W for distances up to 500 feet diameter in an office environment and up to 3000 feet diameter in factories or open plan offices indoors. For line of sight building-to-building communications, the system can achieve distances of 6 miles. With microcell architecture, each base station is capable of handling up to 1 Mbps. The ARLAN 655 and 670 are complete wireless network interface cards that are mounted inside a PC, workstation (WS), or other device. They provide the same functionality as a conventional LAN adapter card and can support multiple topologies in conjunction with the network operating systems.

10.3 WLAN Standards

All standards for WLANs employ unlicensed bands. There are two approaches that can be used to regulate an unlicensed band. One approach is based on a standard to allow different vendors to communicate with one another using a set of interoperable rules. IEEE 802.11 and ETSI's RES 10, HIPERLAN, follow this approach. In the second approach, a minimum set of rules or "spectrum etiquette" is established to allow terminals designed by different vendors to have a fair share of the available channel frequency–time resources and coexist in the same band. This approach does not preclude the first approach. This has been pursued by WINForum. In a coexisting environment, a vendor can be interoperable with another vendor by using the same protocol and transmission scheme.

The three major standard activities for WLANs are: IEEE 802.11, WINForum, and HIPERLAN. IEEE 802.11 developed a standard for DSSS, FHSS, and infrared light technology using the ISM bands as the radio channel. The HIPERLAN standard is aimed at the 5.2- and 17.1-GHz bands in the European countries. The WINForum's goal is to obtain a PCS band for unlicensed data and voice applications and to develop a spectrum etiquette for them.

10.3.1 IEEE 802.11

IEEE 802.11 addresses the physical and media access (MAC) protocol layers for peer-to-peer and peer-to-centralized communications topologies using DSSS or FHSS over radio or infrared light technology. Both SS systems operate in the 2.4- to 2.4835-GHz ISM band. This band has been selected over the 902- to 928-MHz and 5.725- to 5.85-GHz ISM bands because it is widely available in most countries. In the 2.4- to 2.4835-GHz band, more than 80-MHz bandwidth is available that is suitable for high-speed data communication. Also, implementation in this band is more cost effective as compared with the implementation in frequencies that are higher. IEEE 802.11 supports DSSS with binary phase-shift keying and QPSK modulation for data rates of 1 and 2 Mbps, respectively, as well as FHSS with GFSK modulation and two hopping patterns with data rates of 1 and 2 Mbps. For DSSS, the band is divided into five overlapping 26-MHz subbands centered at 2.412, 2.442, 2.470, 2.427, and 2.457 GHz, with the last two overlapping the first three. This setup provides five orders of frequency selectivity for the user. It is quite cost effective to improve the transmission reliability in the presence of interference or severe frequency selective multipath fading. For FHSS, the channel is divided into 79 subbands, each with a 1-MHz bandwidth, and three patterns of 22 hops are user options. A minimum hop rate of 2.5 hops/second is assigned to provide slow frequency hopping, in which each packet is sent in one hop. If a packet is destroyed, the following packet is sent from another hop for which the channel condition would be different. This approach provides a very effective time–frequency diversity and takes advantage of a retransmission scheme to provide a robust transmission. The IEEE 802.11 standard avoids rigid requirements and leaves room for vendors to maneuver in the following areas:

- **Multiple physical media.** FHSS and DSSS radio, as well as infrared light; additional media may be approved in the future.
- **Common MAC layer regardless of physical layer.** All IEEE 802.11-compliant WLANs use CSMA/CA algorithm similar to Ethernet's CSMA/CD MAC layer.
- **Common frame format.** Frames including headers and error protection fields are the same, regardless of whether the attached wired LAN is 802.3 Ethernet or 802.5 token ring; the access point handles conversion of 802.11 frames to wireline frame format.
- **Multiple on-air data rates.** 1 or 2 Mbps, with possibility of higher rates in the future.
- **Power limit.** A maximum power of 1 W (or +30 dBm), as mandated by the FCC; there is no minimum power requirement, which leaves open the possibility of low-power implementations.

The standard defines the basic media and configuration issues, transmission procedures, throughput requirements, and range characteristics for WLAN technology. The standard focuses more on access applications that involve the use of personal digital assistants (PDAs) and portable PCs rather than trunk applications (see figs. 10.1 and 10.2). Trunk applications use wireless as part of the enterprise backbone for transmitting data from building to building, whereas access applications allow users of portable PCs, PDAs, and other wireless devices to tap into corporate LANs from anywhere in an office or on a factory floor.

The radio transmitter in each user end station is always listening for activity on the WLAN. If one end station is transmitting, another will not. The system has a preset time-out to block a user from dominating the network, avoid unnecessary transmission collisions, and allow priority traffic through. This is the function of the CSMA/CA access control